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An FWHT Based FeatureMarking Scheme for Non-repudiate Speech Communication

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***Abstract*—Audio watermarking is the technique that confirms the authenticity or integrity of every audio communication by hiding relevant information in specified areas of the original signal. It has got great importance in the present due to the overwhelmed use of digital media. Watermarking an audio signal helps in guaranteeing copy protection, copyright protection, authorship proof and rightful ownership. Intention of this work is to develop a watermarking scheme that supports authenticity of the signal by labeling it with the Mel-frequency cepstral coefficients (MFCC) that implicitly reveals the identity of its owner.**

***Index Terms*—Audio Watermarking, FeatureMark, Datamatrix Code, FWHT, Non-repudiation**

I. Introduction

ADVANCES in digital technology have led to widespread use of digital communication in various areas including government, legal, banking and military applications. This in turn has increased the reproduction and re-transmission of multimedia data through both legal and illegal channels. However, the illegal usage of digital media causes a serious threat to the content owner's authority or proprietary right. Thus, today's information driven society places utmost importance on authenticating the information that is sent across various communication channels. In the case of digital audio communication schemes these disputes may be the denial of authorship of the speech signal, denial of sending or receiving the signal, denial of time of occurrence etc. Incorporating non-repudiation services in this context guarantees the occurrence of a particular event, the time of occurrence as well as the parties and the corresponding information associated with the event.

Typically, a non-repudiation service should produce cryptographic evidence that guarantee dispute resolution. In other terms, the service should hold relevant information that can achieve the goals against denying their presence or participation. Development of a non-repudiation service should have a service request, in the sense that, the parties involved should agree to utilize the service as well as to generate necessary evidence to support their presence. Evidence of this scheme should be transferred to the other party for the purpose of verification and storage. Separate evidence should be available for the originator as well as the recipient by considering the fact that, any one will not gain any extra benefit from this service and to ensure that the concept of fairness is applied. Timeliness and confidentiality are the other features of a non-repudiation service. [1], [2], [3], [4], [5], [6], [7], [8], [9], [10]

Developing a non-repudiating voice authentication scheme is a challenging task in the context of audio watermarking. Our aim is to suggest a digital audio watermarking scheme that ensures authorized and legal use of digital communication, copyright protection and copy protection that helps to prevent such disputes. Audio watermarking is the term coined to represent the insertion of a signal, image or text of known information in an audio signal in an imperceptible form. The embedded watermark should be robust to any signal manipulations and can be unambiguously retrieved at the other end.

Presently different audio watermarking methods are available, most of them inclined towards copyright protection and copy protection. This was the motivation for the key notion to develop a speaker verification scheme that can guarantee non-repudiation services and the proposed scheme is its outcome. The theme non- repudiation provides immense importance to the audio watermarking scheme proposed in this work.

Comparison with the existing schemes revealed that the proposed scheme is more appropriate in terms of imperceptibility and robustness characteristics. It also ensures the integrity of each member participating in the system and makes it very imperative in the context of signal authentication. The main idea behind this work; realization of a non-repudiation service; is achieved in such a way that the speaker in the communicating group cannot subsequently deny their participation in the communication due to the signal-dependent dynamic watermark.

A theoretical background on Hadamard Transform can be obtained from the papers [11], [12], [13] and [14]. Mel-frequency cepstral coefficients introduced by Davis and Mermelstein in 1980's are treated as the best parametric representation of the acoustic signals employed in the recognition of speakers and have been the state-of-the-art ever since. The schemes proposed in [15], [16], [17] demonstrate audio watermarking as well as audio steganographic techniques in the cepstral coefficients. Extraction of MFCC is understood from the work of [18].

II. Walsh Analysis

Fast Fourier transform or FFT is a common method for evaluating the transform-domain spectra of a digital

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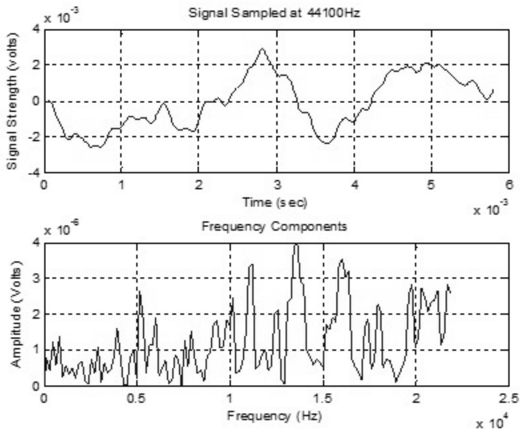

Fig. 1. Amplitude-Frequency Plot

(acoustic) signal. In order to work with this, the time-domain signals must be segmented into finite-length blocks of sampled data called frames. Then we need to evaluate the length of the frames in samples which is evaluated in units of time using the sampling rate.

Walsh transform, a discrete analog of the Fourier transform is employed in this scheme towards the mark embedding process [19], [20], [21], [22], [23]. Due to the fact that Walsh functions and its transforms are naturally more suited for digital computation, an effort is made to gradually replace the Fourier transform by Walsh-type transforms. In this work, the digital watermark is embedded into the original voice signal by applying fast Walsh transform on it and the MFCC features from the audio signal is used in the generation of FeatureMark or the audio watermark. During the transmission process, the FeatureMarked voice signal may suffer various signal manipulations including noise addition, silence addition, echo addition, re-sampling, re-quantization, low-pass filtering, band-pass filtering and other de-synchronization attacks such as amplitude variation, pitch shifting, random cropping, time-scale modification. We adopted Walsh transform for embedding the watermark assuming that in the case of random functions the Walsh power spectra are slowly convergent and many Walsh transform components contain approximately equal signal power. Replacing such coefficients might not degrade the signal quality. Amplitude Vs Frequency plot is represented in figure 1 and the Walsh spectrum obtained by applying the FWHT function on the time-domain signal is represented in figure 2.

After embedding watermark into the selected Walsh coefficients, its inverse function is employed to cancel the changes. During the transmission process, voice signal may suffer some signal manipulations and de-synchronization attacks. The embedding coefficients are selected in such a way that it could hold the robustness nature of the signal. Walsh transforms yields an order of the basis vectors with respect to their spectral properties. Robustness feature of the watermark can be achieved by selecting the Walsh coefficients in such a way that detection and reconstruction of the embedded watermark does not end in any degradation of the data. [24], [25], [26]

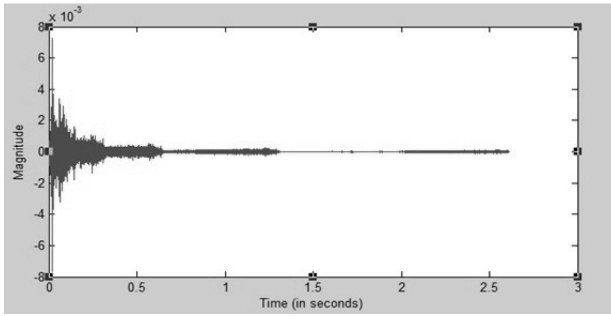

Fig. 2. Walsh Spectrum

Inverse Walsh transform on the transform domain data recovers the original time-domain signal. Inverse Fast Walsh Hadamard Transform (IFWHT) is the function employed towards this process. The "symmetric flag" helps to nullify the numerical inaccuracies, in other words to zero out the small imaginary components and to recover the original signal. The original time-domain signal and the recovered time-domain signal behave almost identically and does not reveal any difference while playing the audio.

III. Proposed Scheme

Normally a speech signal is non-stationary but for short-time duration it appears to be stationary which results from the fact that the glottal system cannot change immediately [18], [27], [28], [29], [30]. Therefore in speech processing it is often advantageous to divide a signal into frames to achieve its stationary nature.

In the proposed scheme, original speech signals are decomposed into a set of overlapping and non-overlapping frames. This is done because the spectral evaluation of a signal is reliable if it is stationary. That is, the region should be short enough for the signal characteristics to be uniform or approximately constant. To achieve this we employed frames with duration of 10 to 25 ms and a frame rate of 100. Obtained frames have sharp edges towards its start and its end and to tone down these edges we employed Hamming windowing technique using the MATLAB's signal processing tool box.

Analysis and characterization of an audio content is performed by audio feature extraction [31], [32], [33]. Some applications that need well-organized feature extraction are Auditory Scene Analysis, Steganography and Watermarking, Content-based Retrieval, Indexing and Fingerprinting etc. As mentioned in [34], the key to extract strong features that characterize the complex nature of audio signals is to identify their discriminatory subspaces. [35], [36], [37], [38], [39], [40]

In the proposed system, feature extraction (figure 3) is done by performing Fourier transform on the signal, where Fourier analysis decomposes the sampled signal into its fundamental periodic components such as sines and cosines. An existing Fast-Fourier transformation such as Cooley, Turkey algorithm maps the given time space into

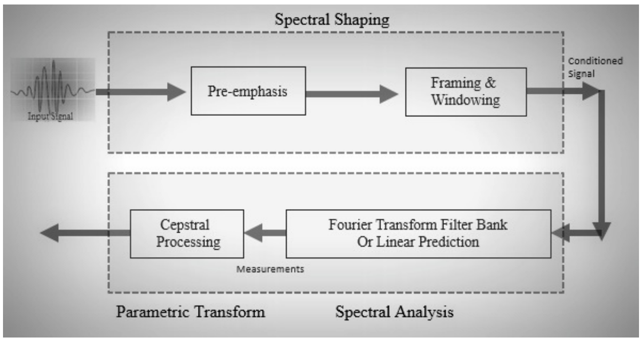

Fig. 3. Feature Extraction

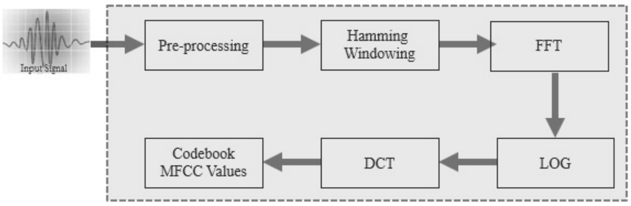

Fig. 4. MFCC Feature Extraction

its corresponding frequency space. Transforming the signal from its time-domain to its frequency-domain is important in the context of audio signals [41].

The computable characteristics of the time-domain signals which are not related to the human perception are extracted and used. The features employed include the Mel-frequency cepstral coefficients. The Mel-Frequency Cepstral Coefficients are restricted to 20 which results in a data matrix of 20×256 coefficients. Embedding such a huge data volume in the audio will expose the furtiveness of the system henceforth vector quantization is implemented resulting in a data matrix of 20×1 coefficients. The steps involved in extracting the MFCC values are depicted in figure 4.

Towards the extraction of the MFCC values, a codebook function for each input signal is performed. This function opens corresponding speech signal and generate a codebook of size 13×16. The size of the codebook can be adjusted by altering the dimension mentioned in the procedure. For each FFT bin, the exact position in the filter bank is identified to find the original frequency response which is preceded by the inversion of filter bank center frequencies. Then identify the integer and fractional sampling positions. Subsequently, the actual processing is started, in which each chunk of data is windowed with Hamming windowing, shift these windows into an FFT order and calculate the magnitude of the FFT. FFT data obtained are converted to the corresponding filter bank outputs and obtain its base 10 log values which is then processed with the cosine transformation in order to reduce its dimensionality. Accuracy of this evaluation procedure is done by the reconstruction of the original filter bank outputs. It involves multiplying the cepstral data by the transpose of the original DCT matrix. Once it is done, FFT bins are combined to make the original signal and it works as expected because the DCT matrix is carefully scaled to be orthonormal.

Watermark Preparation:

Encoding data as its 1-D and 2-D form can be achieved with Barcodes and Datamatrix codes respectively. A data carrier represents data in a machine readable form; used to enable automatic reading of the Element Strings. In this scheme we employed Datamatrix code [42], [43], [44], [45] as its watermark and uses the signal dependent perceptual feature such as MFCC.

A Datamatrix code is the two-dimensional matrix representation which encodes text or numeric data. It can be represented as a square or rectangular pattern with varying arguments of black and white cells which is according to the information to be encoded. The length of encoded data depends on the number of cells in the matrix.

This scheme also employs an encryption method towards the preparation of the FeatureMark which is mentioned below:

$$y = x' \uplus_k \tag{1}$$

y is obtained by inserting $x' mod 7$ at the k^{th} position of x', where $x' = x_1 x_2 ... x_n x_{n+1} ... x_{n+k}$ and $0 \leq k \leq m$ and m = n + k

This guarantees the security of the values that constitute the data code to a good extent. The values obtained are arranged in a specific order to submit as input to an online Datamatrix code generator. Obtained Datamatrix code (figure 5) encodes all the quantized feature vectors of the recorded voice signal. The watermark in the proposed scheme is also stated as the FeatureMark.

Synchronization Code Generation: Synchronization code embedding is a crucial issue associated with all audio watermarking schemes. Detection of watermark bits gets easy by embedding the synchronization code bits in the signal. Any variation in synchronization results in false detection and any spatial or transform domain modifications cause the detector lose its synchronization. Therefore it is suitable to use proper synchronization algorithms based on robust synchronization code. In the proposed approach, a 16-bit Walsh code is embedded in front of the FeatureMark to locate its position [42], [46], [47].

Walsh code is defined as a set of N codes, denoted W_j , for $j = 0, 1 ... N-1$, which have the following properties:

- W_j takes on the values -1 or +1
- $W_j[0]$ = 1 for all j
- W_j has exactly j zero crossings, for $j = 0, 1 ... N-1$
- Each code W_j is either even or odd with respect to its mid-point
- The Walsh code used in this scheme is a 4×4 matrix

Employing the Matlab for the generation of 4×4 Walsh code gives the following matrix:

$$\begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 \end{bmatrix} \tag{2}$$

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In order to embed these bits into the audio signal, we have read the matrix row wise and then calculated its auto-correlation function. The result obtained is a 16-bit sequence and is employed as the synchronization code for this scheme.

A. Embedding Method

Embedding module in this scheme functions through different levels: Synchronization code embedding, FeatureMark embedding.

The First step associated with the embedding module is the segmentation where the original speech signal is divided into a set of segments and then each segment to two sub-segments. Secondly, with the spatial watermarking technique, the Walsh code bits are embedded into the sub-segments. After embedding the synchronization code, the subsequent segments are transformed using the Fast Walsh Transform.

Let $V = v(i), 0 \leq i < Length$ represent a host digital audio signal with Length samples. $FM = FM(i,j), 0 \leq i < M, 0 \leq j < N$ is a binary image to be embedded within the host audio signal and $FM(i,j) \in 0,1$ is the pixel value at $(i,j), S = s(i), 0 \leq i \leq L_{syn}$ is a synchronization code with L_{syn} bits, where $s(i) \in 0,1$.

1) Synchronization Code Embedding: Robust and transparent nature of audio watermarking scheme is achieved by embedding synchronization code bits embedded in it. In this scheme we embedded the Walsh code into the time-domain samples as described below:

- Each audio segment is divided into sub segments having n samples
- Consider the first audio segment $A_{10}(k)(k = 0,1,2...L)$ and its sub segments
- Take four consecutive samples in this segment, let it be s_1, s_2, s_3, s_4 then
- If $s_1 + s_2 > s_3 + s_4$
  - If $s_1 > s_2$
    insert 0 into s_1
  - else
    insert 0 into s_2
- else if $s_3 > s_4$
  - insert 1 into s_3
    else
  - insert 1 into s_4
- else if $s_1 + s_2 = s_3 + s_4$
  - replace with 1 or 0

2) FeatureMark Embedding: In the FeatureMark embedding scheme, Arnold transform is applied to the generated FeatureMark image in order to dissipate its pixel space relationship. It helps in achieving improved robustness of the extracted FeatureMark image. Scrambled image thus obtained is converted into a 1-dimensional sequence of 1s and 0s (binary digits) in order to embed the bits accurately into the 1-dimensional audio signal.

Let $FM = FM(i,j), 0 \leq i \leq M, 0 \leq j \leq N$ represents the original FeatureMark image. Applying Arnold transform results in a scrambled structure which can be represented as

$FM_1 = FM_1(i,j), 0 \leq i \leq M, 0 \leq j \leq N.$

Then scrambled structure will be converted into a sequence of 1s and 0s as follows:

$FM_2 = fm_2(k) = FM_1(i,j), 0 \leq i \leq M, 0 \leq j \leq N, k = i \times N + j, fm_2(k) \in 0,1$

Now, FeatureMark bits are mapped into each signal frames by performing the Fast Walsh transform on the signal. The embedding scheme employs the following condition towards mapping of each FeatureMark bits.

- For the next sub-segment consecutive to the SYNC embedded segment
  - Calculate the magnitude and phase spectrum of each sub-segment using the FWHT : Let $f_1, f_2...f_n$ be the magnitude coefficients obtained
  - Sort the magnitude coefficients in ascending order of their values: Let it be $f'_1, f'_2...f'_n$
  - Energy entropy of each frame is then calculated using the following equation 3
$$I_j = -\sum_{i=1...k} \sigma_i^2 \log_2 \sigma_i^2 \qquad (3)$$
  - Take four consecutive coefficients from this segment, let it be f_1, f_2, f_3, f_4 then
    * If $f_1 + f_2 > f_3 + f_4$
      1) If $f_1 > f_2$
         insert 0 into f_1
      2) else
         insert 0 into f_2
    * else if $f_3 > f_4$
      1) insert 1 into f_3
         else
      2) insert 1 into f_4
    * else if $f_1 + f_2 = f_3 + f_4$
      1) replace with 1 or 0
    * Insert the watermark bits into these coefficients by using the following condition:
      1) For embedding a 1 to f'_n, make $f''_n = f'_n + \alpha$
      2) For embedding a 0 to f'_n, make $f''_n = f'_n - \alpha$, where $\alpha = 0.001$ is a scaling factor
    * Perform the same sequence on the subsequent segments till all the watermark bits are embedded

After embedding all the watermark bits into the signal, the signal should be reconstructed by inverse FWHT tranform and also by combining the set of overlapping and non-overlapping window-frame series.

B. FeatureMark Detection Scheme

The Blind FeatureMark detection scheme functions as a two-step process where synchronization code detection followed by the FeatureMark detection. Synchronization code detection involves identifying the presence of the Walsh code in the signal segments/sub-segments. Since the Walsh code bits are embedded in the time-domain, we can

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Fig. 5. Encrypted Datamatrix Code

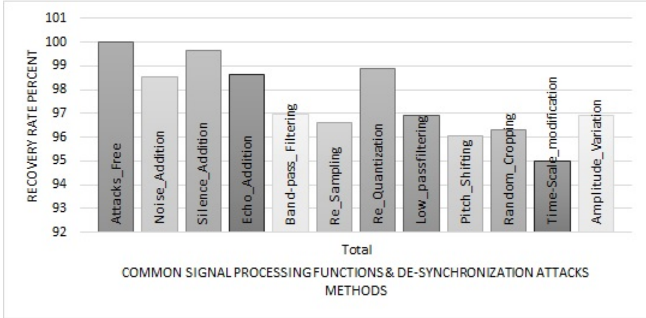

Fig. 6. Average Recovery Rate

directly check the presence of these bits in the sub-segments. Presence of FeatureMark bits are determined by evaluating the spectrum of each frame by Fast Walsh Transform. Then mark bits are fetched out of the signal and are arranged in a matrix of order $M' \times N'$ to finalize the FeatureMark.

FeatureMark bits are extracted by considering the fact that the bits present in between two synchronization codes constitute the embedded FeatureMark (in the case of Repeat Embedding). Extraction of FeatureMark bits also follow the same conditions stated in the embedding module. Use of synchronization code helped us to extract the FeatureMark bits directly out of the marked signal.

C. Experimental Results

Various experiments conducted to evaluate the performance of the proposed scheme endorse its imperceptibility and robustness characteristics. Malayalam speech signals were primarily used for the test. Matlab version R2009b and Music Editor are the basic tools used for inducing common signal manipulations and desynchronization attacks in the sample. Signal duration vary for each signal and in the proposed work, it is in the range of 2-300s. From the experiments conducted, it is obvious that the embedding scheme works fine with most of the samples. An example of the FeatureMark generated for this scheme is shown in figure 5.

Transparency tests were conducted based on a 5-point grade scale [48] and this scheme was excellent in terms of imperceptibility nature.

Robustness Tests: Robustness tests evaluate the FeatureMark detection accuracy against common signal manipulations or de-synchronization attacks. The tests that we carried out are detailed below:

- Measure of strength of watermarking scheme against common signal processing functions such as noise addition, silence addition, echo addition, re-sampling, re-quantization, low-pass filtering & band-pass filtering.
- Measure of strength of watermarking scheme against de-synchronization attacks such as amplitude variation, pitch shifting, cropping & time-scale modification.

Performance of the proposed FeatureMarking scheme is confirmed by assessing the BER values. Average of the recovery rate obtained for some signals are plotted in figure 6.

In this way, the watermark bit error rate and the recovery rate are evaluated to confirm the robust nature of the employed watermarking scheme. Recovery rate is calculated using the equation $(1 - BER) \times 100\%$ that denotes the detection and reconstruction of the embedded watermark without any failure. The BER values obtained and recovery rate shows that the proposed scheme can be applied with the real-time applications involving audio/speech watermarking.

IV. CONCLUSION

Intent of this work is to shape a watermarking scheme that supports legitimacy of the audio signal. The evaluation of the proposed scheme by various research tests reveals the fact that the proposed system had made its benchmark in audio watermarking. The proposed audio watermarking scheme that works with FWHT as its fundamental notion guarantees non-repudiate authentic speech communication. This scheme achieves non-repudiation service to a great extent by employing Mel-frequency cepstral coefficients (MFCC), the best parametric representation of the acoustic signals employed in the recognition of speakers. The scheme devises Walsh code as its synchronization code to improve the robustness nature of the scheme. The system is proficient of assuring copy protection, copyright protection, authorship proof and can even hold the time of occurrence as well as particulars of the parties associated with the event.

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