

A novel method of speech watermarking using reflection coefficient and singular value decomposition in Pisarenko method

Rina Anna Simon and Ajith V Pillai
Department of Applied Electronics,
Rajagiri School of Engineering & Technology,
Kochi, India.
{rinasion, [ajithpillai87](mailto:ajithpillai87@yahoo.com)}@yahoo.com

Abstract— Embedding of data signals in speech has undergone only limited study, but attacks on speech data are quite common. So we propose a method for transmitting message data in speech signal so that it's cryptographically safe. In this method, Discrete Wavelet Transform (DWT) is used to extract the approximation part of host speech signal. The watermark signal is embedded in approximation part of host signal using secret key. The speech signal is non stationary thus linear prediction technique is used to model the speech. The reflection coefficient obtained from linear prediction technique is used as the secret key. The watermark signal along with secret key is embed using Quantization Index Modulation (QIM) technique, into our host signal. At the receiver end, the watermark signal is estimated from the watermarked signal using Pisarenkos' Decomposition method. In order to estimate the watermark signal frequency, adaptive thresholding is done on the watermarked signal using Power Spectrum of the secret key.

Keywords- Reflection Coefficient, Pisarenkos' Decomposition method, DWT, QIM, LPC

I. INTRODUCTION

Speech watermarking is the method of embedding digital signal which include speech, image, video signal in the speech signal. Digital watermarking has been used for speech, as well as speaker, authentication. The watermark is useful information obtained from user which is then embedded either using linear or nonlinear methods. In linear methods, the host signal is linearly combined with embedded signal which is modulated by PN sequence in the case of spread spectrum techniques. In non linear methods, the embedded information is first modulated by sequences of indices and then quantize the host signal using sequence of quantizers. The embedding method is such that it does not cause serious degradation to the host signal. In this paper, a novel method of speech watermarking using QIM method is proposed.

In this method, Discrete Wavelet Transform is used to extract the approximation part of speech signal. It has been already proved that DWT is used to separate voiced and unvoiced regions. Daubechies 10 preserves more perceptual information than other wavelets for speech signal. Embed the

signal along with the secret key, which is the reflection coefficient obtained from LPC analysis, into the host signal. Enhanced Pisarenko's method is used to estimate the frequency from noise watermarked signal.

II. LPC ANALYSIS

Watermark signal energy must be increased as much as possible to speech signal. The speech signal is narrowband signal and whose spectral characteristics changes continuously. The watermark signal must be embedded in perceptually relevant areas. Linear Prediction technique has proven to be highly effective method in modeling speech signal.

In the linear prediction, linear system is modeled as

$$H(Z) = \frac{G}{1 - \sum_{k=1}^P a_k Z^{-k}}$$

Where G is gain and a_k is filter coefficient to be estimated. The LPC coefficients are evaluated by Levinson Durbin recursive technique.

The short term autocorrelation is given by

$$R\alpha=P$$

where R is matrix of autocorrelation coefficients , α is vector of coefficient is vector of autocorrelation coefficient
The filter coefficient α can be computed by using Levinson Durbin algorithm. The reflection coefficient can be computed by

$$rc(i) = r(i) - \sum_{j=1}^{i-1} \alpha_j^{i-1} r(i-j) \quad 1 \leq i \leq P$$

The coefficient rc is called reflection coefficient. The reflection coefficient is more robust to coefficient quantization than predicted coefficients .The reflection coefficient is embedded as secret key along with watermark signal.The bandwidth expansion is performed by multiplying LPC coefficient with a factor ρ .

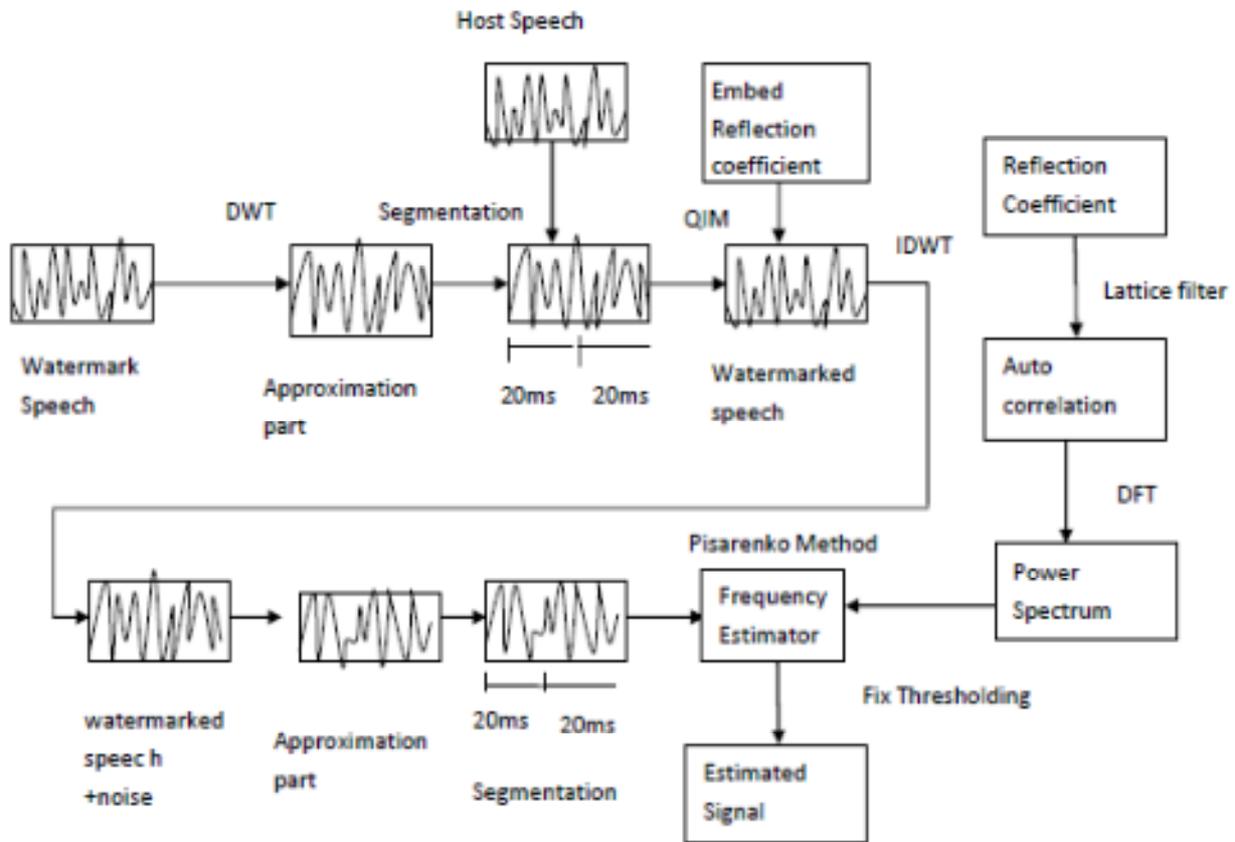


Fig1: Functional Block diagram of Speech Watermarking

$$\alpha' = \alpha\rho$$

ρ must be between 0 and 1 to ensure stability

III. WATERMARKING DETECTION

Secret key are transmitted along with watermarked signal to the Receiver. At the receiver, signal obtained is

$$s(n) = \sum_{n=1}^p w(n) + n(n)$$

Where $s(n)$ is output signal, $w(n)$ is watermarked signal, $n(n)$ is noise signal.

It is possible to estimate the frequency from the peaks of Spectrum. If the received signal is represent as a sum of exponential, then by Pisarenko's Method of frequency estimation we can estimate the frequencies in watermarked signal. Pisarenko's Method uses Eigen decomposition method for frequency Estimation. The disadvantage of using Eigen decomposition is that it can apply to certain class of matrix. So here we use Single Value decomposition. $s(n)$ can be represented as sum of exponentials.

$$s(n) = \sum_{k=1}^p A_k \exp(jw_k n) + n_k(n)$$

The autocorrelation is given by

$$R = R_s + R_n$$

where R_s is the autocorrelation of watermarked signal and R_n is the autocorrelation of noise signal.

$$\text{where } A_k^2 = p_k$$

Let

$$V = [1, e^{jw_k}, e^{jw_k^2}, e^{jw_k^3}, \dots, e^{jw_k(M-1)}]$$

Decompose the autocorrelation of watermark signal using single value decomposition.

$$R_s = V\Lambda U^H$$

Thus,

$$\begin{aligned} R_r &= R_s * R_s^* \\ &= (V\Lambda U^H) * (V\Lambda U^H)^* \\ &= V(\Lambda^* \Lambda) V^* \end{aligned}$$

Where R_r represent resulting Autocorrelation, V represent the Eigen Vector and Λ represent the Eigen values.

$$\Lambda = \text{diag}[p_1, p_2, \dots, p_p]$$

Let $\lambda_1, \lambda_2, \dots, \lambda_M$ be the eigen values and u_1, u_2, \dots, u_M be the normalized eigen vectors

Since R_r is off rank p

$$\lambda_{p+1}, \lambda_{p+2}, \dots, \lambda_M = 0$$

Thus,

$$R_r = \sum_{i=1}^p \lambda_i u_i u_i^H$$

u_1, u_2, \dots, u_p be the principle eigen vector.

In order to model noise signal along with watermarked signal

$$R_n = \sigma_w^2 I$$

The eigen values are denoted by

$$\mu_1, \mu_2, \dots, \mu_M$$

where the first p values

$$\mu_i = \lambda_i + \sigma_w^2 \text{ be the Eigen values}$$

It can be written as

$$R_r = \sum_{i=1}^p (\lambda_i + \sigma_w^2) u_i u_i^H + \sum_{i=p+1}^M (\sigma_w^2) u_i u_i^H$$

Thus it is divided into signal subspace and noise subspace.

In Pisarenko's harmonic decomposition uses orthogonality of two subspaces. Thus noise subspace is spanned by vector U_m which is orthogonal to signal subspace

$$\sum_{k=0}^{M-1} U_{m,k} e^{-jkwi} = 0$$

and hence its $M-1=p$ lies on unit circle which correspond to frequency. Hence frequency is estimated.

The received signal is separated into detail and approximation part by using daubechies 10 wavelet. The secret key which is transmitted along with watermarked is used to produce autocorrelation sequence using lattice filter. Then take the Fourier Transform of autocorrelation sequence to get the Power Spectrum of the water mark signal. The estimated Frequency which contain both host and watermark frequencies. Thus power spectrum is used to set the threshold for obtaining the watermark frequencies from the watermarked signal. This thresholding is used for each segments. If the value above the threshold then it is confirm that the watermark signal get modified otherwise it is correct watermark.

IV. PROPOSED EMBEDDING ALGORITHM

Input : Speech signal, Secret key

Output : Watermarked signal

Step 1: Obain secret key

Step 1.1: Segment the watermark speech signal in the order of 20ms.

Step1.2: Linear Prediction of speech signal is perform then the Coefficients are predicted from Levinson Durbin algorithm.

Step1.3: The by product of Levinson Durbin algorithm is reflection coefficients. This coefficient can be used as secret Key.

Step2: Apply discrete wavelet transform on host signal.

Step 2.1: Apply daubechies 10 wavelet(db10) on the host signal.

Step2.2: Extract the Approximation.

Step 3: Embedding watermark signal along with Secret key using QIM technique.

Step 3.1: Segment the host and watermark signal in the order of 20ms.

Step3.2: Compare the spectrum envelope of both watermark and host signal.

Step3.3: The changes is embedded using QIM technique.

Step3.4: Same procedure to all segment.

Step4: Inverse IDWT is performed.

V. PROPOSED EXTRACTION ALGORITHM

Input : Watermarked signal, secret Key

Output : Alarm for modified/correct data, watermark signal

Step 1: Perform daubechies ten wavelet(db10). Obtain approximation part.

Step 2: Segment the watermarked signal in 20ms

Step3: Using the secret key, autocorrelation of watermark signal is calculated using lattice filter.

Step 4: Take the Fourier Transform of autocorrelation sequence to get Power spectrum

Step 5: Frequency estimation of Watermarked signal is done using Pisarenko's Method.

Step6 : The estimated Frequency contains both host and watermark frequency.

Step7: The thresholding is set by using the Power Spectrum.

If the frequency above this thresholding is discarded because it may be host signal frequency or modified watermark signal.

VI. CONCLUSION

The new method for speech watermarking is presented. In this method, a new method called Pisarenko's Method is used. The frequency masking commonly found in all existing methods can be limited by using this technique. Another advantage of this method is adaptive thresholding, which is used so that both the watermark signal and presence of noise can be detected. It is computationally efficient in real time application.

REFERENCES

- [1] Brian Chen And Gregory W. Wornell, "Quantization Index Modulation Methods for Digital Watermarking and Information Embedding of Multimedia
- [2] Qiang Cheng, Jeffrey Sorensen, "Spread spectrum signalling for speech watermarking,"
- [3] Johnson Ihieh Agbinya, "Discrete wavelet transform in speech processing" IEEE TENCON –Digital signal processing Application
- [4] Martin Hagmuller, Horst Hering, Andreas Kropfl and Gernot Kubin, "Speech Watermarking For Air Traffic Control.
- [5] Libin Cai, Ronghui Tu, Jiyang Zhao, Member, IEEE, and Yongyi Mao, "Speech Quality Evaluation: A New Application of Digital Watermarking," IEEE Transactions On Instrumentation And Measurement, Vol. 56, No. 1, February 2007
- [6] Oscar T.-C. Chen, Member, IEEE, and Wen-Chih Wu, "Highly Robust, Secure, and Perceptual-Quality Echo Hiding Scheme," IEEE Transactions On Audio, Speech, And Language Processing, Vol. 16, No. 3, March 2008
- [7] David J. Coumou, Member, IEEE, and Gaurav Sharma, Senior Member, IEEE, "Insertion, Deletion Codes With Feature-Based Embedding: A New Paradigm for Watermark Synchronization With Applications to Speech Watermarking," IEEE Transactions On Information Forensics And Security, Vol. 3, No. 2, June 2008
- [8] Konrad Hofbauer, Student Member, IEEE, Gernot Kubin, Member, IEEE, and W. Bastiaan Kleijn, Fellow, IEEE, "Speech Watermarking for Analog Flat-Fading Bandpass Channels," IEEE Transactions On Audio, Speech, And Language Processing, Vol. 17, No. 8, November 2009
- [9] Srdjan Stankovic, Senior Member, IEEE, Irena Orovic, Student Member, IEEE, and Nikola ˇ Zaric, Student Member, IEEE, "An Application of Multidimensional Time-Frequency Analysis as a Base for the Unified Watermarking Approach," IEEE Transactions On Image Processing, Vol. 19, No. 3, March 2010
- [10] Mathieu Parvaix, Student Member, IEEE, Laurent Girin, and Jean-Marc Brossier, "A Watermarking-Based Method for Informed Source Separation of Audio Signals With a Single Sensor" IEEE Transactions On Audio, Speech, And Language Processing, Vol. 18, No. 6, August 2010
- [11] Lawrence R. Rabiner and Ronald W. Schafer, "Introduction to Digital Speech Processing" Vol. 1, Nos. 1–2 (2007) 1–194_c 2007 L. R. Rabiner and R. W. Schafer
- [12] Tae Hong Park "Introduction to Digital Speech Processing" computer musically speaking
- [13] Rainer Martin "Statistical Methods For The Enhancement Of Noisy Speech" International Workshop on Acoustic Echo and Noise Control (IWAENC2003), Sept. 2003, Kyoto, Japan
- [14] V. F. Pisarenko, The retrieval of harmonics from a covariance function, Geophysics J. Roy. Astron. Soc. 33 (1973), 347–366.