Available online at www.ijiere.com
International Journal of Innovative and Emerging Research in Engineering
e-ISSN: 2394 – 3343 p-ISSN: 2394 – 5494

Voice Transmission by Wavelet Packet Modulation

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ABSTRACT:
The main aim of this project is to transmit voice by using new method i.e., wavelet packet modulation and demodulation technique. It will replace the old modulation techniques and also overcome most of the disadvantages of them. It is proposed to implement this project in Battle Field Management and many other industrial places where we need an effective voice transmission. For this, the system requires a real time recording device to record the voice message, it is then modulated and demodulated by using Wavelet Packet Transformation, the modulation and demodulation techniques is carried out by using MATLAB software. The carrier used is AWGN of 10-15db, as it consists of all types of environmental noise.

Keywords: wavelet packet modulation, wavelet packet demodulation, real time voice recording, IDWPT, DWPT, Denoising, data compression.

I. INTRODUCTION

Quality of Voice in the telecommunication industry is an important field from of all common technical and economic points of view. Voice contains two frequencies one is higher frequency components and other is lower frequency components. Generally all the information is retained in lower frequency components and higher frequency components are neglected as it contain less information part. Audio coding and quality assessment techniques have generally the need to account for the properties of the human auditory system. Drawing on research on hearing, mathematical models may be constructed that reproduce key properties such as loudness perception and masking. This measurement is important for companies maintaining the transmission lines of any kind (metallic, optical, wireless etc.)[1].

II. JUSTIFICATION AND MOTIVATION

This paper compares techniques for speech signal noise reduction based on discrete wavelet transform techniques. Earlier discrete wavelet transform (DWT) was studied to improve the system robustness against the noise of 0dB. Also, state of the art wavelet denoising techniques have been very successfully applied to image noise reduction. However it has not yet been widely used to solve the speech signal noise reduction problem[3], as few publications in wavelet in comparison to enormous STFT papers. Because both Fourier and wavelet transforms are linear and noises are additive, the STFT solutions should applicable to the wavelet domain.

Fourier transform works well for wide sense stationary signals, but is not much effective for use in non-stationary signals, because of its hidden localization information. This was improved by the short time FT but it also leads to the spectral distortion due to the window properties.

The motivation to use wavelet as a possible alternative for noise reduction is to explore new ways to reduce computational complexity and to achieve better noise reduction performance. Firstly, because the wavelet transform may not require overlapped windows due to the localization property, the same filter could process less data than before. Secondly, wavelet filter does not correspond to time domain convolution; so that shift-invariant is not preserved but the Fourier domain filters can still be extended to the wavelet domain, because they are derived according to the statistical properties of spectral components. Thirdly, reason could also be that there are many different wavelets and various wavelet transform combinations. Therefore, a possibility for a better wavelet transform is great[2].

III.RELATED WORK

A mathematician comes up with a good idea, develops a concrete theory, faces great opposition from other prominent figures in the area, but continues to work nevertheless. Then come in engineers and physicists, reformulate and modify that theory to make it more accessible, and eventually that idea becomes a standard tool for many researchers in many fields of their researches. History of mathematics and engineering is full of such stories, but the similarity between two particular ones is quite striking. In 1807, Joseph Fourier, a French mathematician, discovered that all periodic functions could be expressed as a weighted sum of basic trigonometric functions. His ideas faced much criticism from Lagrange, Legendre and Laplace for lack of mathematical rigor and generality, and his papers were denied publication. It took Fourier more than 15 years to convince them and publish his results. Over the next 150 years his ideas were expanded and generalized for non-periodic functions and discrete time sequences[5].

The FFT algorithm, devised by Cooley and Turkey in 1965 placed the crown on Fourier transform, making it the king of all transforms. Since then Fourier transforms have been the most widely used, and often misused, mathematical tool in not only electrical engineering, but disciplines requiring function analysis. Following a remarkably similar history of development, the wavelet transform is rapidly gaining popularity. With applications ranging from pure mathematics to virtually every field of engineering, from astrology to economics, from oceanography to seismology, wavelet transforms are being applied successfully to such areas where no other transform has ever been applied[4].

IV. PROPOSED METHOD

Wavelet packet transform is a simple generalization of a wavelet transform. All wavelet packet transforms are calculated in a similar way. So we have focused initially on the Haar wavelet packet transform, which is the easy to describe. The Haar wavelet packet transform is usually referred to as the Walsh transform.

<u>Discrete Wavelet Transform [6]:</u>
DWT is designed from the multi-resolution analysis that
decomposes the given signal space into an approximate space, *V*, and detail spaces, *W*, as shown in equation (1).

$$V_{j+1} = W_j \oplus V_j = W_j \oplus W_{j-1} \oplus V_{j-1} \quad (1)$$

where W_j is the orthogonal complement of V_j in V_{j+1} and $\oplus$ represents the orthogonal sum of two subspaces V_j and W_j, which are constructed by orthonormal scaling functions ($\varphi_{j,k}$)and orthonormal wavelet functions ($\psi_{j,k}$) respectively. Scaling function and wavelet function are obtained as

$$\begin{aligned}\phi_{j,k}(t) &= 2^{j/2}\phi(2^j t - k) = \sum_l h_{l-2k}\phi_{j+1,k}(t)\\ \psi_{j,k}(t) &= 2^{j/2}\psi(2^j t - k) = \sum_l g_{l-2k}\phi_{j+1,k}(t)\end{aligned} \quad (2)$$

With low-pass filter, $h_{l-2k} = (\varphi_{j,k}, \varphi_{j+1,l})$ and high-pass filter, $g_{l-2k} = (\psi_{j,k}, \varphi_{j+1,l})$. *()* means inner product. Using these functions, DWT of a given signal "f", provides scaling coefficients and wavelet coefficients. The scaling coefficient and wavelet coefficient at the *j*th level *k*th time is computed by

$$c_{j,k} = \langle f, \phi_{j,k}\rangle = \sum_l h^*_{l-2k}\langle f, \phi_{j+1,l}\rangle = \sum_l h^*_{l-2k} c_{j+1,l} \quad (3)$$

$$d_{j,k} = \langle f, \psi_{j,k}\rangle = \sum_l g^*_{l-2k}\langle f, \phi_{j+1,l}\rangle = \sum_l g^*_{l-2k} c_{j+1,l} \quad (4)$$

Figure showing the 2-level analysis part of the DWT and its frequency separation property is as shown.

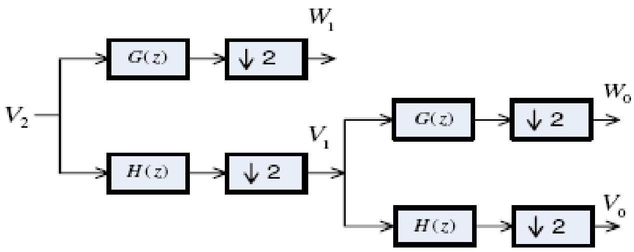

Figure 1: level analysis part of the DWT[6]

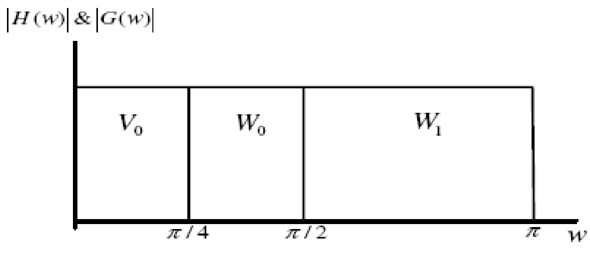

Figure 2: Frequency separation of 2-level analysis part of the DWT in case of using ideal filter bank[6]

Haar System[7]:

The Haar orthogonal system begins with Φ (t), the characteristic function of the unit interval given by

$$\Phi (t) = x_{[0, 1]}(t) \quad (5)$$

It is clear that Φ (t) and Φ (t - n), $n \neq 0$, $n \in Z$ are orthogonal as their product is zero. It is also clear that $\{\phi (t – n)\}$ is not a complete orthogonal system in L^2 (R) since its closed linear span V_o consists of 2 piecewise constant functions with possible jumps only at the integers. The characteristic function of (0, 1/2), with a jump at 1/2 can not have a convergent expansion.

To include more functions we need to consider the dilated version of ϕ (t) as well, $\phi (2^m t)$ where $m \in Z$. Then by making a change of variable we see that

$\{2^{m/2} \phi (2^m t – n)\}$ is an orthonormal system. V_m denote its closed linear span since any function in L^2 (R) may be approximated by a piecewise constant function f_m with jumps at binary rational, it follows that is dense in L^2 (R). Thus the system $\{\phi_{mn}\}$

$$\phi_{mn} (t) = 2^{m/2} \Phi (2^m t - n) \qquad \bigcup_n V_m (6)$$

is complete in L^2 (R), but, since ϕ (t) and ϕ (2t) are not orthogonal so it is not an orthogonal system. Therefore we must modify it somehow to convert it into an orthogonal system.

But the solution is simple; let $\Psi(t) = \phi (2t) - \phi (2t - 1)$. Now everything works; $\{\psi (t - n)\}$ is orthonormal system, and ψ (2t - k) and ψ (t - n) are orthogonal for all k and n. This allow us to deduce that $\{\psi_{mn}\}_{m,n\in Z}$, where

$$\psi_{mn} (t) = 2^{m/2} \psi (2^m t – n), \quad (7)$$

is a complete orthonormal system in L^2 (R). This is the Haar system; the expansion of $f \in L^2$ (R) is as follows

$$f(t) = \sum_{m=-\infty}^{\infty} \sum_{n=-\infty}^{\infty} \langle f, \psi_{mn} \rangle \psi_{mn} (t), \quad (8)$$

The ϕ (t) is usually called as the scaling function in wavelet terminology while Ψ(t) is the mother wavelet[7].

$$f_m(t) = \sum_{k=-\infty}^{m-1} \sum_{n=-\infty}^{\infty} \langle f, \psi_{kn} \rangle \psi_{kn} (t).$$

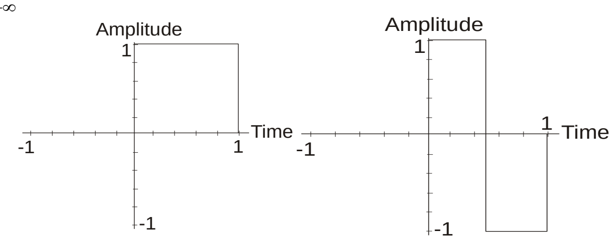

Figure3: (a) The scaling function and (b) mother wavelet for the Haar system[7].

Walsh Function[7]:

The Rademacher functions are orthogonal system on (0, 1) and were obtained by combining the Haar functions by simply adding them at a given scale. Walsh functions take the sums and differences of the Haar functions to obtain a complete system. We define

$$W_o (t) := \phi (t), \quad (10)$$
$$W_1 (t) := \Psi (t), \quad (11)$$
$$W_2 (t) := \Psi (2t) + \Psi (2t - 1), \quad (12)$$
$$W_3 (t) := \Psi (2t) - \Psi (2t - 1), \quad (13)$$
$$W_{2n} (t) := W_n (2t) + W_n (2t - 1), \quad (14)$$
$$W_{2n+1} (t) := W_n (2t) - W_n (2t - 1). \quad (15)$$

These Walsh functions also belong to the wavelet subspaces of the Haar system:

$W_0 \in V_0$, $W_1 \in W_0$, W_2 , $W_3 \in W_1$, W_4 , W_5 , W_6, $W_7 \in W_2$,...........W_{2m},W2m_{+1},.............,$W_2{}^{m+1}{}_{-1} \in W_m$,........ (16)

Notice that these defining relations (16) are exactly the same as those in the two dilation equations of the Haar system,

$$\phi (t) = \phi (2t) + \phi (2t – 1), \quad (17a)$$
$$\Psi (t) = \phi (2t) - \phi (2t - 1). \quad (17b)$$

Since all functions defined by (17a) are orthogonal to all defined by (17b), it follows that W_{2n} and W_{2n+1} are orthogonal. Also if W_n and W_m are orthogonal then W_{2m}, W_{2m+1}, …, W_{2m+1-1} are orthogonal in W_n. All of these functions have support contained in [0, 1], so $\{W_n\}$ are an orthogonal system in L^2 (0,1). Also, there are exactly 2^m Haar functions in W_m whose support lies in [0, 1] and therefore the Walsh functions in W_m form a basis of this space. Since the Haar functions are complete in L^2 (0, 1) and so are the Walsh functions[7].

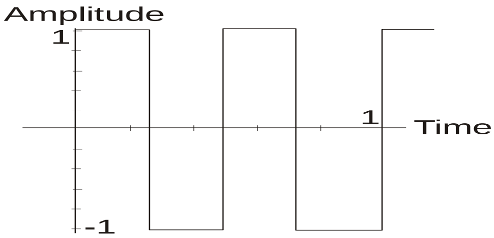

Fıgure4: One of the Rademacher functions[7]

Also the Haar wavelet packet transform is usually referred to as the Walsh transform. A Walsh transform is calculated by performing a 1-level Haar transform on all subsignals present, both trends and fluctuations[7].

V. BLOCK DIAGRAM

Following block diagram shows the flow of procedure followed from recording of voice to addition of white Gaussian noise and finally the recovering of original voice signal.

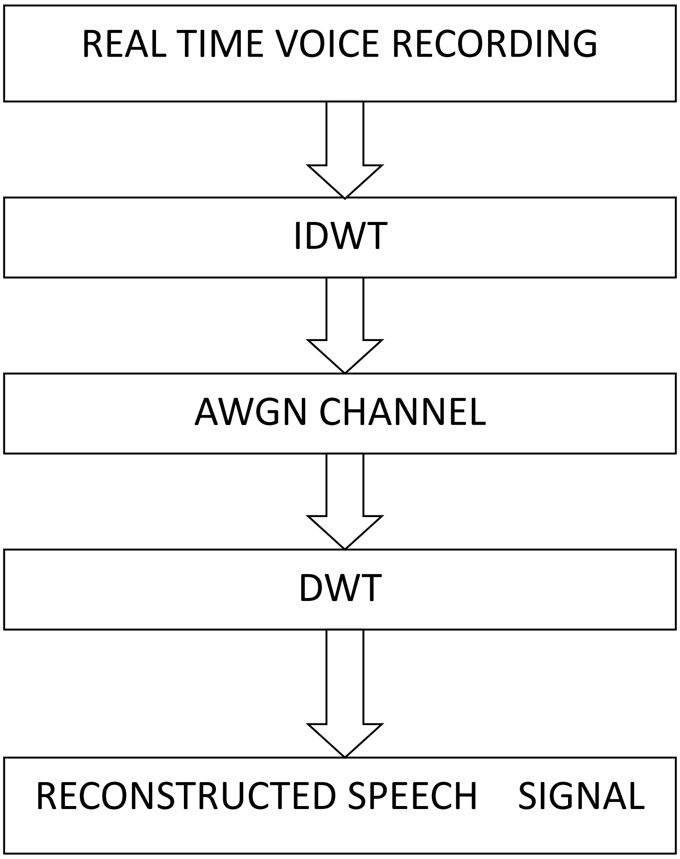

Figure5: Block diagram of voice transmission by wavelet packet modulation

The methodology has been broken down into the following steps

1. Real time Voice Recording

To record the voice we will require a mic and a recorder. This can be done by an inbuilt microphone in laptops or Headphone can be connected to PC. The voice which is recorded for a short period of time is stored. In this project we have recorded for 5-10 secs.

2. IDWT

The next step is to use the voice data which is stored in the PC/Laptops for IDWT (Inverse Discrete Wavelet Transform), It converts the voice signal into spectrums, which is easier to transmit.

3. AWGN Channel

AWGN is used as a channel noise during modulation of voice data. Here we are using 15dB of AWGN noise along with the voice signal. AWGN noise is chosen because it consists of all types of environmental noise.

4. DWT

The transmitted signal is received by the receiver is then undergo DWT and the original signal is retraced successfully. The received signal contains negligible amount of noise or very less noise, thus it makes this process a very effective and efficient to use.

VI. RESULTS

First the voice is recorded for few secs using the recording function. Recorded voice is then added with the white Gaussian noise of 15 db. The signal is then demodulated at receiver to get back the original signal containing the information equally near to what was transmitted.

At the transmitter side IDWT is performed and at the receiver side DWT is performed.

The waveforms given below are showing the recorded (voice/speech) signal, white Gaussian noise and final recovered signal at the receiver. Final received signal contains less of noise and more of information.

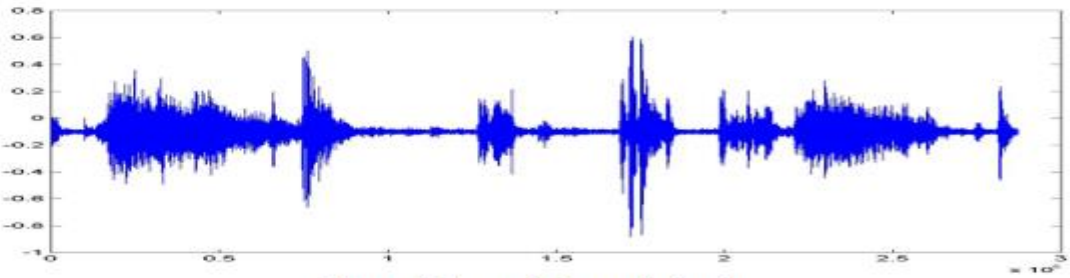

Figure 6: Recorded speech signal

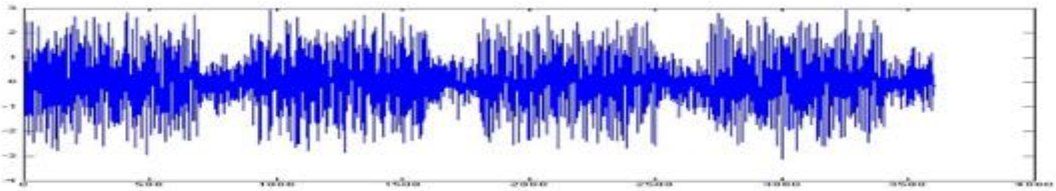

Figure 7: Channel signal(modulated signal)

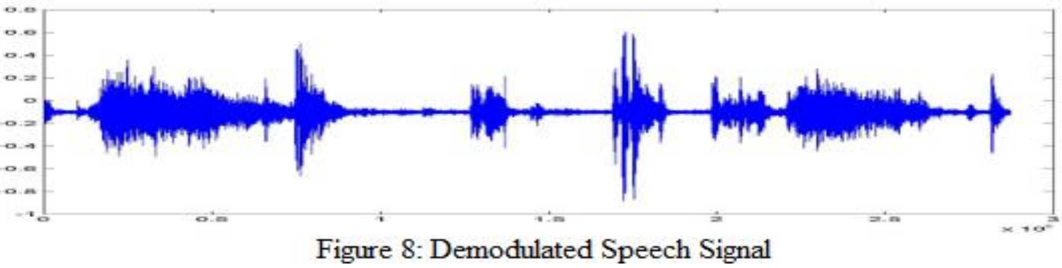

Figure 8: Demodulated Speech Signal

VII. CONCLUSION

In this paper, we presented an overview of research activities dealing with voice transmission by wavelet packet modulation. We used wavelet transformation and related research issues. We introduced three different modules i.e. voice recording, modulation and demodulation of voice. This paper addresses the methodology of transmission of voice using wavelet for more effective purpose of transmission. This technology will provide quick response with less error and less time delays. It is hoped that this work helps to provide less complex techniques and robust system. In the future, we believe that the proposed methodology is flexibly enough to be further improved and proposed to implement this project in Battle Field Management and many other places where we need an effective voice transmission also applied to standard industrial information transmission systems.

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