Investigating the Effect of Error Correcting Codes on the Compressed Speech Signals

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***Abstract*— Data compression algorithms are designed to reduce the size of data so that it requires less bandwidth to be transmitted over data communication channels of limited bandwidth, and error correcting codes are used for increasing the robustness against channel impairments (noise, interference, fading, ..). This paper presents a performance evaluation of a communication system including a Wavelet based speech coder, which is implemented to compress the speech signal size. A significant advantage of using wavelets for speech coding is that the compression ratio can easily be varied, and the obtained signal to noise ratio is extremely high. The compressed speech signal has been converted to a binary and encoded with Error correcting codes. Convolutional and Reed Solomon codes have been used as an error correcting code to detect and correct the errors at the receiver side, and interleaving used with the Convolutional codes in order to increase the capability of error detection and correction. The system has been tested for convolutional codes with two different code rate values (1/2 and 2/3), and these two values are compatible with the type of modulations that have been used (4-QAM and 8-QAM). Finally it has been concluded that adding error correcting codes improves the system performance by approximately 3.7 to 5.9 dBs.**

Keywords-component; Speech compression; Wavelet; Error correcting codes, Convolutional Codes.

I. INTRODUCTION

Speech can be defined as the response of the vocal tract to one or more excitation signals. Compression of signals is based on removing the redundancy between neighboring samples and/or between the adjacent cycles. In data compression, it is desired to represent data by as small as possible number of coefficients within an acceptable loss of visual quality [1].

Speech compression is the technology of converting human speech into an efficiently encoded representation that can be decoded later to produce a close approximation of the original signal. Major objective of speech compression is to represent speech with less or few number of bits with level of quality [2].

Errors are introduced in the compressed speech when passed through the channel. The channel noise interferes the signal. The signal power is also reduced. Hence errors are introduced. The transmission of the data over the channel depends upon two parameters, the transmitted power and channel bandwidth. The power spectral density of the channel noise and these two parameters determine signal to noise power ratio. The signal to noise ratio determine the probability of error of the modulation scheme. For the given signal to noise ratio, the error probability can be reduced further by using coding techniques [3].

Error correction coding is the means whereby errors which may be introduced into digital data as a result of transmission through a communication channel can be detected and corrected based upon received data. Error detection coding is the means whereby errors can be detected based upon received information. Collectively, error correction and error detection coding are error control coding. Error control coding has a significant enabler in the telecommunications revolution, the internet, digital recording, and space exploration. Error control coding is nearly ubiquitous in modern, information-based society. Every compact disc, CD-ROM, or DVD employs codes to protect the data embedded in the plastic disk. Every hard disk drive employs correction coding. Every phone call made over a digital cellular phone employs it. Every packet transmitted over the internet has a protective coding “wrapper” used to determine if the packet has been received correctly [4].

There are a lot of studies about speech compression alone, but there are just few works studying the effect of error correcting codes on these compressed data, such as in [5] Cyclic Coding algorithm for Original and Received Voice Signal at 8 KHz using BER performance through Additive White Gaussian Noise Channel is proposed and it is seen that Cyclic Code reduces the effect of error on transmitted signal caused by AWGN channel.

The goal of this paper is to study and evaluate the effect of the type of error correcting codes such as, Convolutional Codes (CC) and Reed-Solomon (RS) Code on the compressed speech in Additive White Gaussian Noise (AWGN) channel. The speech has been compressed using Discrete Wavelet Transform (DWT).

II. SPEECH CODING

The goal of all speech coding systems is to transmit speech with the highest possible quality using the least possible channel capacity. Speech coding can be seen as a method to compress speech with good voice quality.

Wavelet Based Speech Coding

The speech coding system is based on an orthogonal transform which is the wavelet transform. Wavelet is used to extract many un-useful information of the signal, and truncate them to zero value in the wavelet domain, so long series of zeros are obtained after the truncating process, by encoding these zeros using a significant type of coding technique which is called the run-length encoding the size of the speech signal is compress. The run-length coding program which has been used in this work encodes successive equal amplitude coefficients not only zeros which gives more compression ratio up to 3% difference with other programs which encode only zeros [6,7].

Wavelets work by decomposing a signal into different resolutions or frequency bands, and this task is carried out by choosing the mother wavelet and computing the Discrete Wavelet Transform (DWT). Signal compression is based on the concept that selecting a small number of approximation coefficients (at a suitably chosen level) and some of the detail coefficients can accurately represent regular signal components. The most important factor that has an effect on the results is the mother wavelet chosen as the coif 5 since it has a high vanishing moment, so it gave the best results. Also level of decomposition into wavelet domain has its big influence on the result and the SNR (level three gives the best compression ratio and signal o noise ratio) [6,7].

III. Error Correction Codes

Undesirable disturbances (noise) can occur across the communication channel, causing the received information to be different from the original information sent. Coding theory deals with detection and correction of the transmission errors caused by the noise in the channel. The primary goal of coding theory is efficient encoding of information, easy transmission of encoded messages, fast decoding of received information and correction of errors introduced in the channel. Error Correction can be handled in two ways:

1- Backward Error Correction: In this scheme, when an error is discovered; the receiver has a back channel to request the sender to retransmit the entire data unit, also known as ARQ.

2- Forward error correction: In this, receiver can use an error-correcting code, which automatically corrects certain errors [8].

Applications that require low latency (such as telephone conversations) cannot use Automatic Repeat request (ARQ); they must use Forward Error Correction (FEC). By the time an ARQ system discovers an error and re-transmits it, the re-sent data will arrive too late to be any good. Applications where the transmitter immediately forgets the information as soon as it is sent (such as most television cameras) cannot use ARQ; they must use FEC because when an error occurs, the original data is no longer available [9].

In this research Convolutional codes with two different code rate (1/2 and 2/3) and Reed-Solomon code with code rate (3/7) have been used as an error correcting codes. Interleaving has been used with the convolutional codes in order to increase the capability of error detection and correction.

A. Convolutional Code

Convolutional codes are linear codes that have additional structure in the generator matrix, so that the encoding operation can be viewed as a filtering - or convolution - operation.

Many telecommunications applications have used convolutional codes because of their ability to deliver good coding gains on the AWGN channel for target bit error rates around 10^{-5} [10]. So it is a powerful and widely used class of codes, which are used in a variety of systems including today's popular wireless standards (such as 802.11) and in satellite communications.

Convolutional codes are often preferred in practice over block codes, because they provide excellent performance when compared with block codes of comparable encode/decode complexity. Whereas block codes take discrete blocks of K symbols and produce there from blocks of N symbols that depend only on the k input symbols, convolutional codes are frequently viewed as stream codes**,** in that they often operate on continuous streams of symbols not partitioned into discrete message blocks [4].

When the encoded information is transmitted over the channel, it is distorted**;** the convolutional decoder regenerated the information by estimating the most likely path of state transition in the trellis. The receiver, of course, does not have direct knowledge of the transmitter's state transitions. It only sees the received sequence of parity bits, with possible corruptions. Its task is to determine the best possible sequence of transmitter states that could have produced the parity bit sequence. This task is called decoding, a decoder that is able to infer the most likely sequence is also called a maximum likelihood decoder. The Viterbi decoder finds a maximum likelihood path through the Trellis [11].

B. Interleaving

Interleaving plays a vital role in improving the performance of Forward Correction (FEC) codes in terms of Bit Error Rate (BER). Interleaving is the process to rearrange code symbols so as to spread burst of errors into random like errors and thereafter FEC techniques could be applied to correct them. In conventional block interleaver, the bits received from the encoder are stored row wise in the interleaver's memory and read column wise as shown in Figure 1, WiMAX uses a special type of block interleaver in which the Interleaver Depth (ID) and pattern vary depending on the code rate and modulation type [12].

C. Reed-Solomon (RS) Code

RS code is a cyclic code. Cyclic codes are a subset of the class of linear block codes that satisfy the following cyclic shift property: if $\mathbf{c} = (c_{n-1}\, c_{n-2} \ldots c_1\, c_0)$ is a codeword of a cyclic code, then $(c_{n-2}\, c_{n-3} \ldots c_0\, c_{n-1})$, obtained by a cyclic shift of the elements of **c,** is also a codeword. That is, all cyclic shifts of **c**

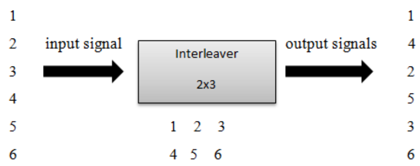

Figure 1: Interleaving process.

are codewords. The RS codes are probably the most widely used codes in practice. It is a special class of nonbinary BCH. RS codes are *t*-error-correcting 2^m –ary BCH codes with block length:

$N= 2^m -1$ symbols (i.e. mN binary digits), where m is any positive integer greater than or equal to 3 and $1 \leq t \leq 2^{m-1} -1$

$N-K = 2t$, where K is the input message length [13].

IV. THE PROPOSED SIMULATED SYSTEM

The proposed simulated systems block diagrams are shown in Figure 2 and 3. The speech information signal (shown in Figure 4) is first compressed using DWT, The DWT coefficients (using coif 5 mother wavelet and level three decomposition), threshold coefficients, and the encoded coefficients are shown in Figures 5-7.

For the system in Figure 2, the compressed speech converted to a digital (binary) signal and then encoded using Convolutional Codes as an error correcting code. Two types of Convolutional Codes have been used, one with a code rate R_c=1/2, and other with R_c=2/3, when:

$$R_c=K/N$$

To increase the error correcting capability of the code, the encoded binary signals have been interleaved using block interleaver. Then, before transmitting the signal via an Additive White Gaussian Noise (AWGN) channel, the signal has been modulated using compatible QAM modulation. As shown in TABLE I, 4-QAM modulation was used when R_c=1/2, and 8-QAM used when R_c=2/3. Actually this was done, because the codeword length of the first type of Convolutional code is 2 (2^2 = 4), and the codeword length of the second type is 3 (2^3 =8).

TABLE I. TYPE OF MODULATIONS AND THE CODE RATE VALUES

Modulation	**Convolutional Code rate (*Rc*)**
4-QAM	1/2
8-QAM	2/3

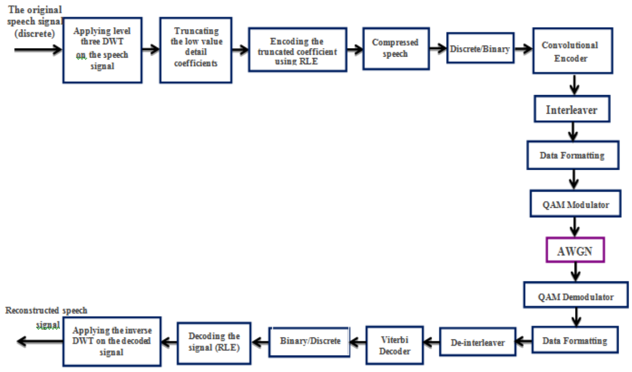

Figure 2: Simulated system block diagram using Convolutional Code with Interleaver.

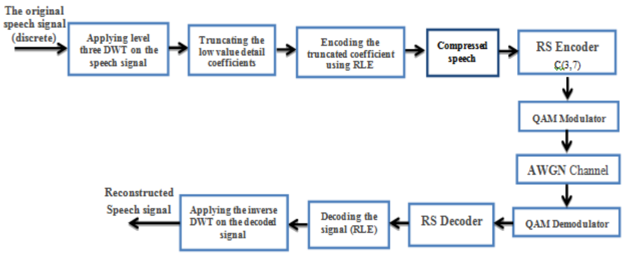

Figure 3: Simulated system block diagram using RS Code.

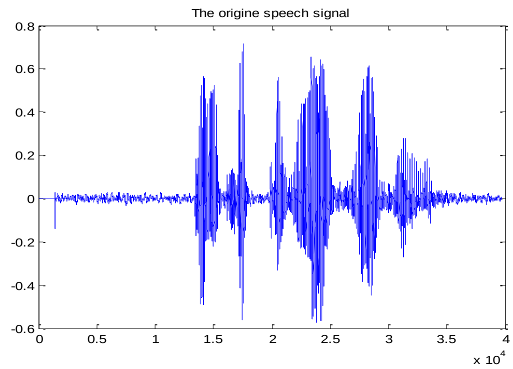

Figure 4: The original speech signal.

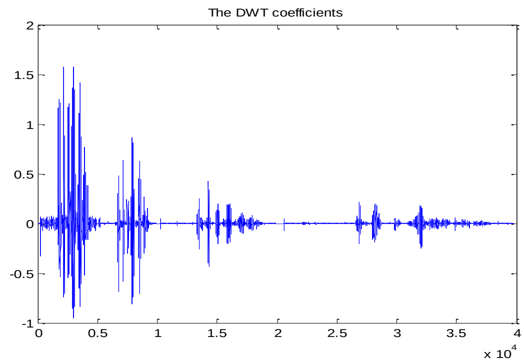

Figure 5: The (level 3) decomposition DWT coefficients using coif5.

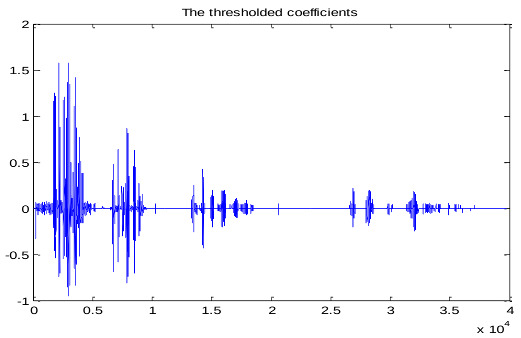

Figure 6: The truncated coefficients.

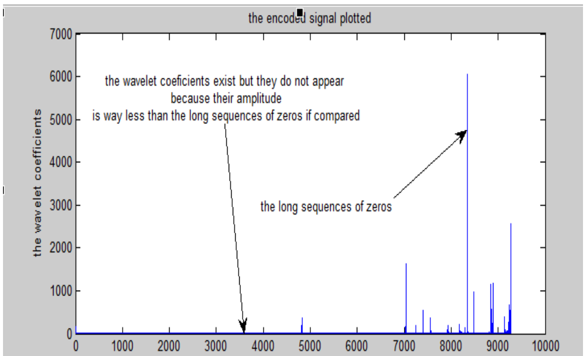

Figure 7: The encoded coefficients.

At the receiver, the received signal demodulated, the digital signal de-interleaved, and then using Viterbi decoder for correcting the errors and decoding the digital signal. Then the compressed speech is decode using RL decoder, and finally applying the inverse DWT on the decoded signal in order to reconstruct the speech signal.
The transmitted compressed speech parameters are as below:
Compression ratio %=(1-(f_a/f_b))*100)
f_b**:** is the file size before compression, and
f_a **:** is the file size after compression.
The compression ratio=78%
SNR=10log$_{10}$ [σ_x^2/ σ_e^2]
σ_x**:** is the mean square of the original speech signal.
σ_e**:** is the mean square difference between the original and the reconstructed speech signals.
The SNR before the transmission process =28 dB.
For the system in Figure 3, the same compressed speech signal has been encoded using different type of error correcting code, Reed-Solomon codes have been used with a code rate (3/7), and 8-QAM modulation has been used with.

V. RESULTS AND DISCUSSIONS

The systems have been simulated using MATLAB. The performances of the proposed systems have been introduced depending on Bit error rate (BER) versus signal to noise (SNR) ratio plots. The system has been evaluated without using any type of error correcting codes, and in this case to achieve 10^{-5} BER, the SNR ratio should be approximately 9.7dB.
For the case of adding Convolutional Code with a code rate (R_c=1/2) and block interleaving, only 3.8dB is needed to achieve 10^{-5} BER, and when increasing the code rate of the Convolutional Code to 2/3, the needed SNR also increased to approximately 6dB to reach the same BER.
When using non-binary Reed-Solomon Code with a code rate 3/7, the needed S/N ratio is 5.1dB to reach 10^{-5} BER.
All the results have been put together in Figure 8.
The reconstructed speech signal for the system with Convolutional error correction is shown in Figure 9. The S/N ratio is 10.16dB and the speech quality is still very good due to the used error correction codes. Transmitting speech signals through AWGN channel without error correction is not efficient, as shown in Figure 10. It is clear that the quality of the signal is not accepted (the S/N is only 4.08 dB), and when listening to the speech signal the speech is not understandable.

VI. CONCLUSIONS

In this paper a compressed speech using DWT has been transmitted over AWGN channel, using different types of error correcting codes, Convolutional Codes with code rates (1/2 and 2/3), RS code with code rate 3/7, and different types of modulation, beside interleaving, which has been used with the Convolutional Codes.
The robustness of using speech coder is of prime importance (a system that gives high compression ratio and signal to noise ratio). This can be obtained by choosing a suitable transform for speech processing which is the wavelet transform, choosing the best mother wavelet for compression which is coif 5, and finally choosing the right decomposition level which is level three.
From the results shown in Figure 8, it could be seen that adding the error correcting codes improves the performance by about 3.7 to 5.9 dBs.
So, it could be concluded that the Convolutional codes and the simple RS codes are a good candidates for correcting errors happening by AWGN channel for compressed speech using DWT.

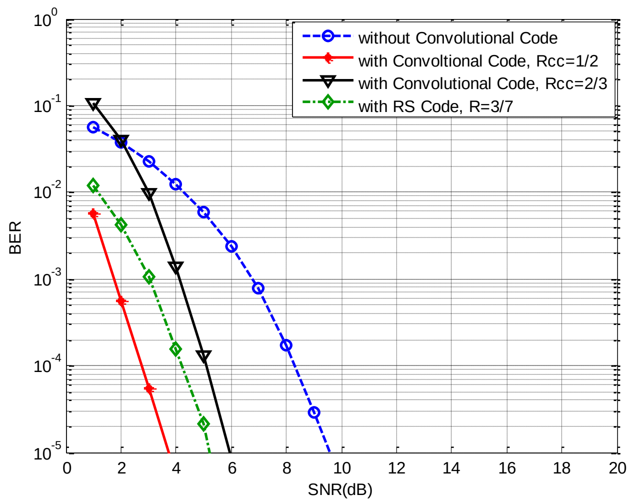

Figure 8: SNR versus BER performance for all the evaluated cases.

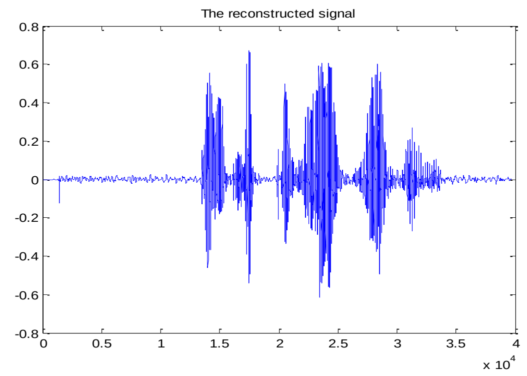

Figure 9: The reconstructed speech signal with error correction.

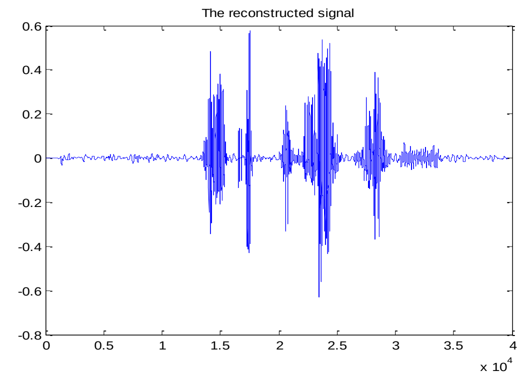

Figure 10: The reconstructed speech signal without error correction.

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