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Performance Analysis of Acoustic Signal in OFDM Using FEC Codes

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Abstract –**This paper addresses the analysis of speech communication over an orthogonal frequency division multiplexing (OFDM) based system and also the impact of Forward Error Correction (FEC) codes namely Reed Solomon encoder Code and Convolution Code on the performance of OFDM wireless communication system for speech signal transmission over both AWGN and fading (AWGN, Rayleigh and Nakagami) channels in term of Bit Error Probability. The simulation has been done under different modulation techniques i.e. BPSK, QPSK and QAM with different code rates.**

Keywords – **AWGN, BER, BPSK, FEC, Nakagami Channel, OFDM, QAM, QPSK, Rayleigh Channel.**

I. INTRODUCTION

The growing demand for wireless multimedia services requires reliable and high-rate data communications over a wireless channel. Ideally, wireless multimedia systems have to be adaptive based on the channel conditions. In this paper, we include the role of the application layer and investigate the use of adaptive modulation in an OFDM system used for transmitting progressively-coded signal with multiple description coding. Specifically, each description is mapped into one of the subchannels of the OFDM waveforms. In most of the literature, such as, temporal coding is used. In this paper, we use a cyclic redundancy check (CRC) to check the validity of each description, and erase all descriptions that do not pass the CRC. Then, Reed Solomon (RS) erasure decoding is used across the descriptions. To achieve minimal signal distortion, we need to optimize the constellation size and code rates jointly. However, due to the complexity of jointly optimizing adaptive modulation and channel coding, the problem is decomposed into two sub problems. First, we decide the constellation sizes to maximize the system throughput prior to RS decoding, then we decide the code rates to minimize distortion. In much of the literature, the same constellation size is used for all the subchannels when applying adaptive modulation for signal transmission. However, we propose adopting different constellations for different subchannels to avoid the problem of overwhelming some of the subchannels by imposing a higher order modulation size than the quality of their channels can sustain. Specifically, two schemes of M-QAM adaptive modulation are considered. The first is a variable rate, fixed power scheme; for each subchannels, a constellation size is assigned which maximizes the system throughput prior to RS decoding, with equal power allocation for all subchannels. The second is a variable rate, variable power scheme, which maximizes the system throughput prior to RS decoding by changing the constellation size and the allocated power at each sub channel.

II. FORWARD ERROR CORRECTION CODING

Forward error correction (FEC) or channel coding is a technique used for controlling errors in data transmission over unreliable or noisy communication channels. The central idea is the sender encodes their message in a redundant way by using an error-correcting code (ECC). The redundancy allows the receiver to detect a limited number of errors that may occur anywhere in the message, and often to correct these errors without retransmission. FEC gives the receiver the ability to correct errors without needing a reverse channel to request retransmission of data, but at the cost of a fixed, higher forward channel bandwidth. FEC is therefore applied in situations where retransmissions are costly or impossible, such as when broadcasting to multiple receivers in multicast. FEC information is usually added to mass storage devices to enable recovery of corrupted data.

The two main categories of FEC codes are block codes and convolutional codes.

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- Block codes work on fixed-size blocks (packets) of bits or symbols of predetermined size. Practical block codes can generally be decoded in polynomial time to their block length.
- Convolutional codes work on bit or symbol streams of arbitrary length. They are most often decoded with the Viterbi algorithm, though other algorithms are sometimes used. Viterbi decoding allows asymptotically optimal decoding efficiency with increasing constraint length of the convolutional code, but at the expense of exponentially increasing complexity. A convolutional code can be turned into a block code, if desired, by "tail-biting".

III. Reed Solomon Codes

Reed–Solomon (RS) codes are non-binary cyclic error-correcting codes invented by Irving S. Reed and Gustave Solomon. They described a systematic way of building codes that could detect and correct multiple random symbol errors. By adding t check symbols to the data, an RS code can detect any combination of up to t erroneous symbols, and correct up to $\lfloor t/2 \rfloor$ symbols. As an erasure code, it can correct up to t known erasures, or it can detect and correct combinations of errors and erasures. Furthermore, RS codes are suitable as multiple-burst bit-error correcting codes, since a sequence of $b+1$ consecutive bit errors can affect at most two symbols of size b. The choice of t is up to the designer of the code, and may be selected within wide limits.

In Reed–Solomon coding, source symbols are viewed as coefficients of a polynomial p(x) over a finite field. The original idea was to create n code symbols from k source symbols by oversampling p(x) at n > k distinct points, transmit the sampled points, and use interpolation techniques at the receiver to recover the original message. That is not how RS codes are used today. Instead, RS codes are viewed as cyclic BCH codes, where encoding symbols are derived from the coefficients of a polynomial constructed by multiplying p(x) with a cyclic generator polynomial. This gives rise to efficient decoding algorithms (described below).

Reed–Solomon codes have since found important applications from deep – space communication to consumer electronics. They are prominently used in consumer electronics such as CDs, DVDs, Blu-ray Discs, in data transmission technologies such as DSL and WiMAX, in broadcast systems such as DVB and ATSC, and in computer applications such as RAID 6 systems.

Original view (transmitting points)

The original concept of Reed–Solomon coding (Reed & Solomon 1960) describes encoding of k message symbols by viewing them as coefficients of a polynomial $p(x)$ of maximum degree $k-1$ over a finite field of order N, and evaluating the polynomial at $n>k$ distinct input points. Sampling a polynomial of degree $k-1$ at more than k points creates an over determined system, and allows recovery of the polynomial at the receiver given any k out of n sample points using (Lagrange) interpolation. The sequence of distinct points is created by a generator of the finite field's multiplicative group, and includes 0, thus permitting any value of n up to N.

Using a mathematical formulation, let $(x_1, x_2, \ldots\ldots, x_n)$ be the input sequence of n distinct values over the finite field F; then the codebook C created from the tuplets of values obtained by evaluating every polynomial (over F) of degree less than k at each x_i is:

$$\boldsymbol{C} = \{(\boldsymbol{f}(\boldsymbol{x_1}), \boldsymbol{f}(\boldsymbol{x_2}), \ldots, \boldsymbol{f}(\boldsymbol{x_n}))\ |\boldsymbol{f} \epsilon \boldsymbol{F}[\boldsymbol{x}], \mathbf{deg}(\boldsymbol{f}) < \boldsymbol{k}\} \quad (1)$$

Where $F[x]$ is the polynomial ring over F, and k and n are chosen such that $1 \leq k \leq n \leq N$. As described above, an input sequence $(x_1, x_2, \ldots\ldots, x_n)$ of $n = N$ values is created as $(0, a^0, a^1, \ldots\ldots, a^{N-2})$, where α is a primitive root of F. When omitting 0 from the sequence, and since $a^{N-1} = 1$ it follows that for every polynomial $p(x)$ the function $p(\alpha x)$ is also a polynomial of the same degree, and its codeword is a cyclic left-shift of the codeword derived from $p(x)$; thus, a Reed–Solomon code can be viewed as a cyclic code. This is pursued in the classic view of RS codes, described subsequently.

As outlined in the section on a theoretical decoder, the original view does not give rise to an efficient

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decoding algorithm, even though it shows that such a code can work.

Classic view (Reed–Solomon codes as BCH codes)

In practice, instead of sending sample values of a polynomial, the encoding symbols are viewed as the coefficients of an output polynomial $s(x)$ constructed by multiplying the message polynomial $p(x)$ of maximum degree $k-1$ by a generator polynomial $g(x)$ of degree $t = N - k - 1$. The generator polynomial $g(x)$ is defined by having $a, a^2, \ldots . a^t$ as its roots, i.e.

$$\boldsymbol{g(x) = (x-a)(x-a^2)\ldots(x-a^t) = g_0 + g_1 x + \cdots + g_{t-1}x^{t-1} + x^t} \quad (2)$$

The transmitter sends the $N-1$ coefficients of $s(x) = p(x)g(x)$, and the receiver can use polynomial division by $g(x)$ of the received polynomial to determine whether the message is in error; a non-zero remainder means that an error was detected [3]. Let $r(x)$ be the non-zero remainder polynomial, then the receiver can evaluate $r(x)$ at the roots of $g(x)$, and build a system of equations that eliminates $s(x)$ and identifies which coefficients of $r(x)$ are in error, and the magnitude of each coefficient's error. (Berlekamp 1984) (Massey 1969) If the system of equations can be solved, then the receiver knows how to modify his $r(x)$ to get the most likely $s(x)$.

Reed–Solomon codes are a special case of a larger class of codes called BCH codes. The Berlekamp–Massey algorithm has been designed for the decoding of such codes, and is thus applicable to Reed–Solomon codes.

To see that Reed–Solomon codes are special BCH codes, it is useful to give the following alternative definition of Reed–Solomon codes [4] Given a finite field F of size q, let $n = q - 1$ and let α be a primitive n^{th} root of unity in F. Also let $1 \leq k \leq n$ be given. The Reed – Solomon code for these parameters has code word $(f_0, f_1, \ldots, f_{n-1})$ if and only if $a, a^2, \ldots, a^{n-k}$ are roots of the polynomial

$$\boldsymbol{p(x) = f_0 + f_1 x + \cdots + f_{n-1}x^{n-1}} \quad (3)$$

With this definition, it is immediately seen that a Reed–Solomon code is a polynomial code, and in particular a BCH code. The generator polynomial $g(x)$ is the minimal polynomial with roots $a, a^2, \ldots, a^{n-k}$ as defined above, and the code words are exactly the polynomials that are divisible by $g(x)$.

IV. Convolutional Coding

A Convolution encoder contains a shift register that offers temporary storage and a shifting process for the input bits and exclusive-OR logic circuits which produce the coded output from the bits currently held in the shift register. In general, k data bits can be shifted into the register at once, and n code bits generated. In practice, it is often the case that $k = 1$ and $n = 2$, giving rise to a rate ½ code. A rate ½ encoder illustrated in figure 1 [10] and this will be used to explain the encoding operation.

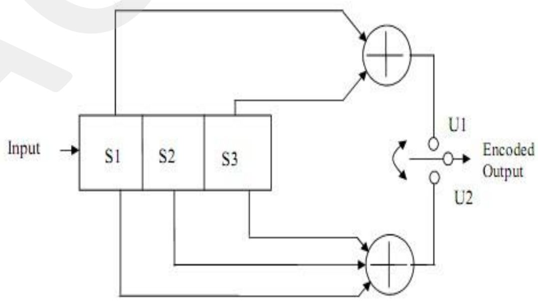

Figure 1: A ½ rated Convolution Encoder [10]

In the beginning, the shift register holds all binary 0s. The input data bits are fed in continuously at a bit rate R_b, and the shift register is clocked at this rate. As the input changes through the register, the rightmost bit is moved out so that there are always 3 bits held in the register. At the end of the message, three binary 0s are devoted, to return the shift to its initial condition. The commutator at the output is switched at twice the input bit rate so that two output bits are produced for every input bit shifted in. At any one time the register holds 3 bits which from the input to the exclusive-OR circuits.

Convolutional codes have been extensively applied to satellite communications. In cellular mobile communication, the channel characteristics is less favourable with burst errors arising from

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multipath (reflection), shadowing of the signal and co-channel interference, but the necessity to achieve coding gain at the adequate target bit error rates yet again dictates that convolutional codes should be used. For the reason that of the hostile channel environment, the voice coders (vocoders) are designed to work well with bit error rate of 10^{-3} and acceptable with error rates well beyond this. Convolutional codes are greatly suitable for AWGN channels [11]. In the case of GSM standard for digital mobile communication, convolutional codes are desirable with interleaving to protect against the channel error burst.

V. System and Channel Models

System Model

For the transmission of progressively-coded signal over an OFDM system with L sub channels, an embedded bit stream is first converted into L descriptions using an FEC-based multiple description coder. Then, Reed-Solomon (RS) encoding is used to code across the descriptions and provide unequal error protection for the multiple descriptions, where the rates of the codes are a non-decreasing function of the level of importance of the data. Lastly, a cyclic redundancy check (CRC) is appended to each description for error detection. Note that the terms description and packet are used interchangeably in this paper.

Coding across the subchannels normally requires a consistent code alphabet. However, this paper proposes to have variable modulation alphabet sizes across the subchannels using adaptive modulation. Hence, a mapping from the modulated symbols to the RS code symbols is needed. For adaptive modulation, the constellation size M is restricted to 2η, where η is an even number varying from 2 to N_b. For $N_b = 6$, the resulting constellation choices are 4-QAM, 16-QAM and 64-QAM.

With adaptive modulation, the number of symbols modulating the subchannels is the same, but the number of bits may vary from sub channel to sub channel. As a GF (210) RS code is adopted, each RS code symbol contains 10 bits. We mapped the modulated symbols to the RS code symbols as shown in figure.3. Five 4-QAM, 16-QAM, and 64-QAM modulated symbols are grouped as one, two, and three RS code symbols, respectively.

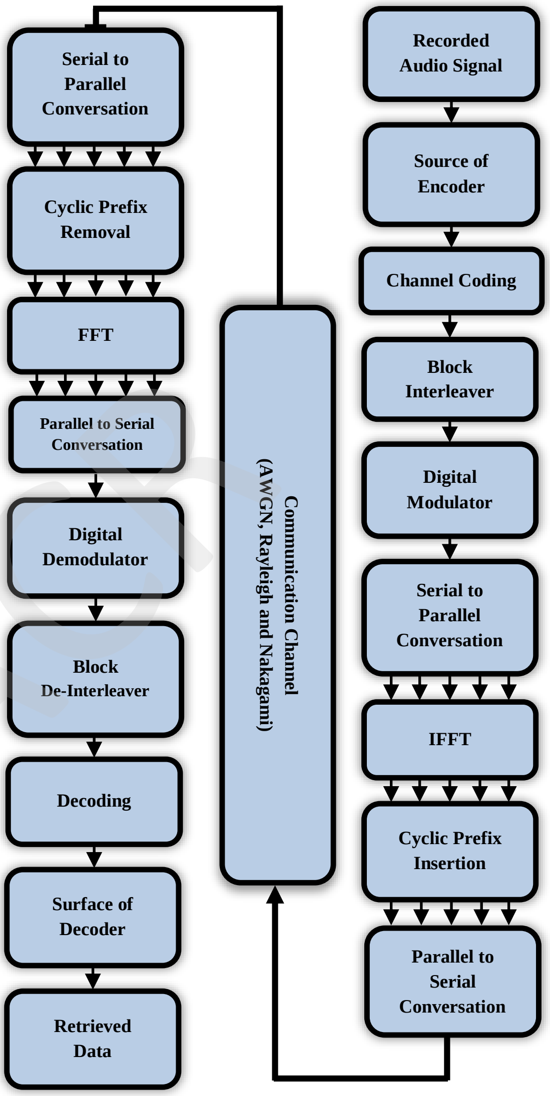

Figure 2: Basic model of OFDM system

Channel Model

In a frequency-selective OFDM channel, the entire frequency band of BT Hz is assumed to be divided into L independent and identically distributed flat fading subchannels, with bandwidth approximately equal to the coherence bandwidth of the channel. Furthermore, slow Rayleigh fading is assumed at each sub channel, such that the fading coefficient remains constant over a packet. We assume neither intersymbol interference nor inter-carrier interference exists.

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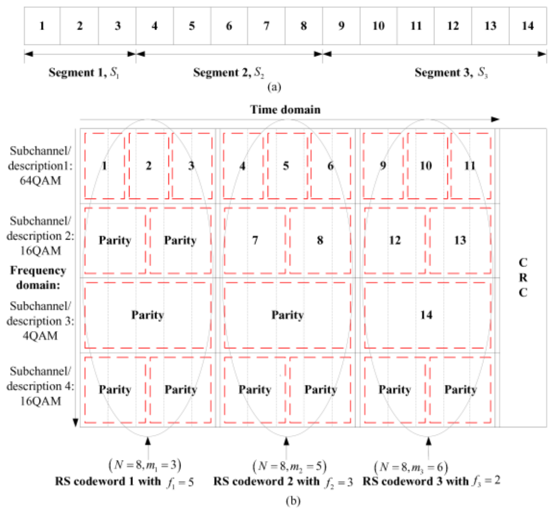

Figure 3: Illustration of converting an embedded bit stream into multiple descriptions with L = 4 descriptions and J = 3 RS code words. (a) An embedded bit stream partitioned into 3 segments. (b) Mapping of the descriptions to RS code words.

VI. Simulation and Results

Simulations are done with MATLAB 2008.

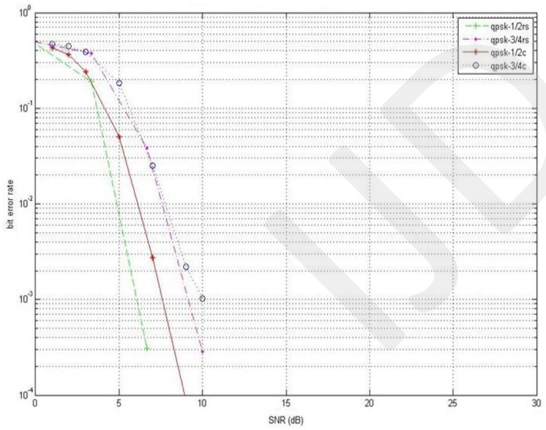

Figure 4: BER v/s SNR graph for m=1

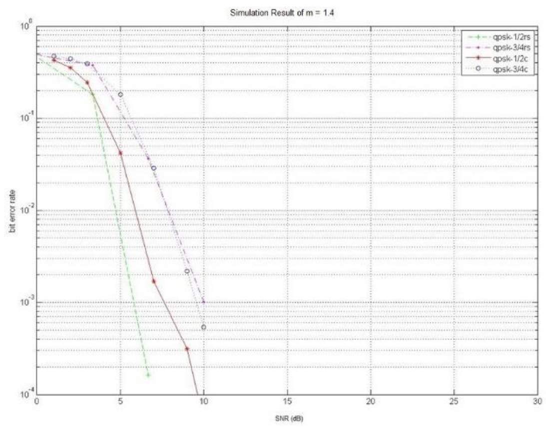

Figure 5: BER v/s SNR graph for m=1.4

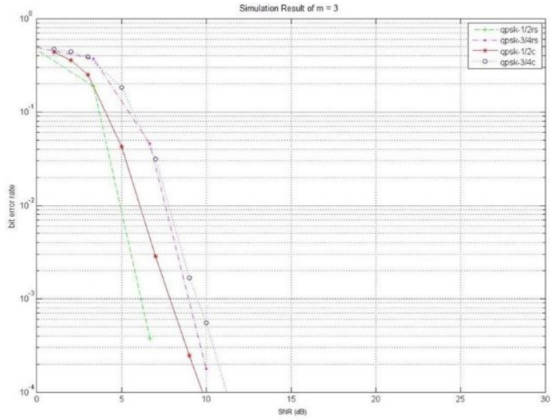

Figure 6: BER v/s SNR graph for m=3

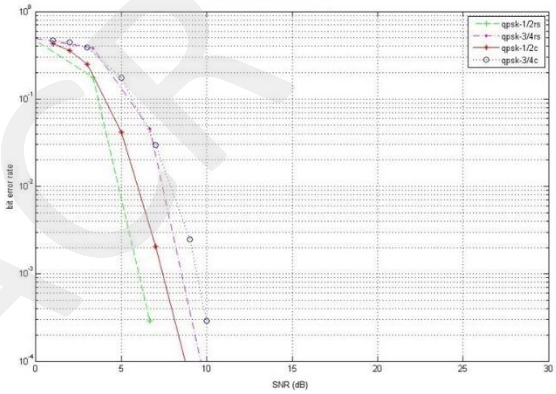

Figure 7: BER v/s SNR graph for m=5

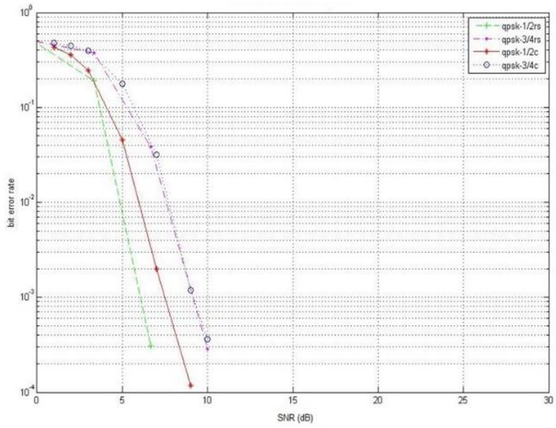

Figure 8: BER v/s SNR graph for m=15

VII. Conclusion

This paper presents an analysis of acoustic signal for OFDM system with the help of FEC codes under AWGN, Rayleigh and Nakagami fading channels. It is found that, as the value of m > 1 increases, the BER performance improves.

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