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A Comprehensive Review on Error Control Coding mechanisms in High Speed Wireless Transmission System

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ABSTRACT

The development of efficient coding techniques and continuous enhancement of those with respect to the accelerated data rate requirements is the need of today, to achieve good quality of the transmission systems without any distortion in the message. The demand is more in the current scenario due to the emergence of large scale high speed data networks for transmitting and processing digital information. A simultaneous development in communication systems and computer technology, and the effective merging of both techniques is required to design efficient systems. The occurrence of errors is a major concerning factor in communication systems which reduce the security as well as the effectiveness of the system. The main focus of the designer is to detect and control these errors, so that the data can be reliably reproduced. There are various error controlling and detecting codes that are used to give a secured and effective output. This paper reviews the requirements for error control coding for next generation wireless.

KEYWORDS: *Random Error, Burst Error, Error control codes, Erasure codes, WiMAX.*

I. Introduction

There is a need for continuous improvement in the wireless transmission technology, to increase the data rate, transmission distance, to reduce bandwidth and transmission error. The IEEE 802.16 set of standards is formed during October 2004 to achieve this objective[1]. This Worldwide Interoperability for Microwave Access (WiMAX) standard is characterized to support high scalability, fast positioning and high transmission rate of up to 75 Mbps for fixed wireless metropolitan access networks (MAN). The standard was upgraded in October 2005 as IEEE 802.16e to include support for mobile access[2]. The 802.16e uses modulation technique of scalable orthogonal frequency division multiplexing (OFDM) and the transmission technique of multiple input and multiple output (MIMO) to get a higher data rate. The other implications of the standard include large coverage, high frequency usage, larger power savings, greater bandwidth support and also provision for good Quality of Service (QoS).

The successful usage of the technologies such as wireless internet, video conferencing, network modeling and RF based Identification depends on the effective deployment of the WiMAX standards. In ideal situation, the IEEE 802.16e is capable of achieving a trans mission rate of 75 Mbps over a range of 112.6 km. But in practice, due to the topographical differences, the standard supports data rate up to 10 Mbps at around 10 km range. There is a further drop in transmission, when the users are on move. The main reason for low data rate at high speed is due to the fading of radio signal and lower range carrier frequency. The data transmission capability is limited at higher speed, owing to the signal noises generated at the receiving end which results in higher rate of erroneous in message.

Shannon proved that the errors in the system can be reduced to a satisfactory level without compromising the information rate through proper encoding. Coding is a technique used to optimize the digital communication system[3]. The signal quality can be increased through error control coding. It is possible to achieve a reliable system performance, which is the need of the hour for the high speed digital systems being used today. The objective of error control coding is to deliver information from a source to a destination with minimum amount of errors.

In this article the stage by stage development of coding theory, types of error occurs in the channel and its controlling strategies, coding and decoding procedures, are discussed in detail.

II. Nature of Errors in Wireless Channel and Its Controlling Strategies

The type of error in a channel depends on the effect of signal noises on the transmissions. In random error channels, which are also known as memoryless channels, the noise disturbs each transmitted symbol independently. The error effects in deep space channels and many satellite channels come under this category. The codes required to overcome these errors are called as random error correcting codes. The other type of error channel is called as burst error channels, in which the noise is dependent between transmissions. The error is found in clusters due to high transition probability within transmissions. The mobile channel and storage medium is an example of burst error channels. The error clusters in mobile channels occur due to signal fading of the multipath transmission and cable lines. The coding required to avoid these types of errors are called burst error correcting codes.

The error control coding strategies are of three types, forward error correction (FEC), automatic repeat request (ARQ) and hybrid coding which is a combination of FEC and ARQ. The FEC codes are used to correct error automatically at the receiver end. The ARQ is a two way system employing error detection and retransmission. The error detection system requires simpler decoding equipment than error correction. In FEC, back channel is not necessary i.e. the retransmission of data can be avoided at the cost of huge spectrum requirement. For wireless channels at high vehicular speeds, the ARQ strategy has limited impact on the effective data rate because the system cannot afford the extra overheads caused by the reactive protocols when the usable data rate is already very low. Because of high error rate, the reactive schemes like ARQ require several attempts to send uncorrupted data to the mobile nodes. Since retransmission of data is not required, the FEC mechanism is most suitable for high speed wireless communication. Merging of FEC and ARQ is a suitable solution in a scenario where error rate is high.

The coding performance is measured in terms of reduction in error rate for a given signal-to-noise ratio (SNR) or as SNR difference at a fixed bit or symbol error rate, the latter being used generally. Coding gain is defined as the difference in value of bit energy to noise ratio (Eb/No) required in attaining a particular error rate with and without coding. The coding gain is obtained at the expense of transmission bandwidth; hence it is the reciprocal of the code rate. Nowadays, in digital communication systems, coding schemes that deliver coding gain of 2 to 8 dB are extensively used. This is made possible due to the significant decrease in the cost of digital hardware [4].

III. History of Error Control Techniques and Decoding Algorithms

Shannon laid the foundation for channel coding in 1948 by establishing the possibility of a reliable communication over a noisy channel [3]. In his 'Cheating channel' example, for a given noisy forward channel of capacity C1, a noiseless feedback channel and a noiseless forward channel of capacity C2, the transmission rate would be C1+C2. This can be achieved by a simple coding procedure which is easy to implement. This is in contrast to the complex codebook procedures followed to transmit over a single noisy channel with low error probability in the absence of feedback.

Shannon thus demonstrated that feedback can simplify coding and decoding but without any improvement in capacity. In case of analog sources, minimum distortion occurs when the source is encoded and sent over a given channel. But it may need very complex encoding. In case of a band limited Gaussian source sent over a Gaussian channel of same bandwidth, no encoding is needed to achieve the minimum distortion. But, if the bandwidths are different, the complexity of encoding reappears.

According to Pete work in 1956 titled Channel Capacity without Coding, the complex coding can be avoided using a simple strategy, if the channel bandwidth are integer multiple of the source bandwidth with some available feedback. On the other hand, the coding complexity reappears when the bandwidth is different from the above mentioned [5]. Pete's results are further analysed by Elias with an approach that simplifies coding with feedback [6]. Kelly [7] in 1956 demonstrated that the channel capacity is independent of any requirement to coding the signals that are to be transmitted over the channel.

Hamming [8] in 1950 developed single error correction block code which is considered the first practical channel code. For decoding block codes, algebraic algorithm, table look-up algorithm and majority logic algorithms are used [4].

Hamming codes are decoded using syndrome table. The disadvantage of the Hamming code is that when the block size increases the performance of the code drops and reaches a point where the relative error correction rate is zero. The examples of cyclic and its related codes are Reed-Solomon (RS) [9], maximal-length, Bose – Chaudhuri – Hocquenhem (BCH)[10], Reed-Muller code, quadratic residue code, projective geometry code, Golay code, difference sets code, Goppa code, Euclidean geometry code and quasi-cyclic codes.

The BCH codes are generalization of the Hamming code and are used for multiple error correction. They are best suited for channels in which errors affect successive symbols independently. But they also suffer with the same short coming as that of Hamming codes. An optimum algorithm is to be used to determine the dimension and minimum distance of BCH codes. Frobenius stable check matrix gives a better estimate of the minimum distance. For decoding BCH codes, Berlekamp's iterative algorithm and Chien Search Algorithms are used.

The RS codes are used widely in many applications due to its powerful non-binary block codes. As the correction of the codes is done at system level, the RS codes are superior in rectifying burst errors. The error locator polynomial is used to locate the errors and the Forney's algorithm [11] is used to find the magnitude of the errors. The error locator polynomial can be found using Berlekamp–Massey (BM) algorithm or Euclid's algorithm. Though the BM algorithm is more efficient [12], the algorithm of Euclid is generally used due to its easy implementation. Then the error locations are identified using Chien search algorithm [13]. The Fig.1 compares the theoretical decoding error probability of RS(127,106) and RS(31,21) in AWGN channel over uncoded system and the coding of around 5.5 dB is obtained for the target rate of 10^{-4}[14]. Also, it is observed that the decoding error probability progressively drops as the E_b /N_0 and code dimension decrease. This reflects the imperfect characteristics of RS codes.

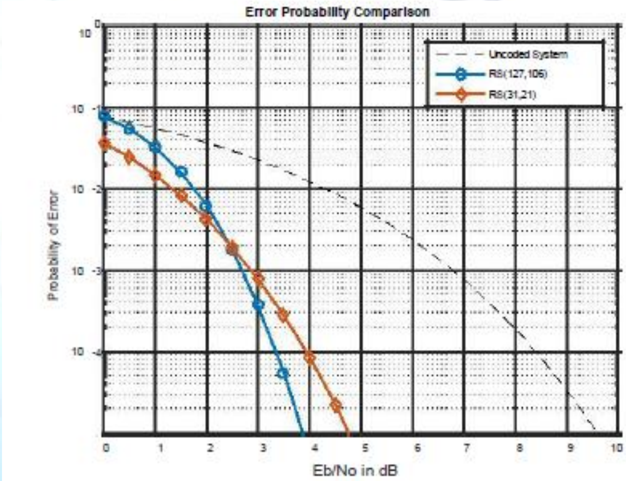

Fig. 1 Decoding Error Probability of system with and without RS codes

The theoretical error performance of RS (127, 63) and RS (127,106) in AWGN channel is compared with Rayleigh fading channel in Matlab environment and the results are shown in the following figure, Fig. 2.

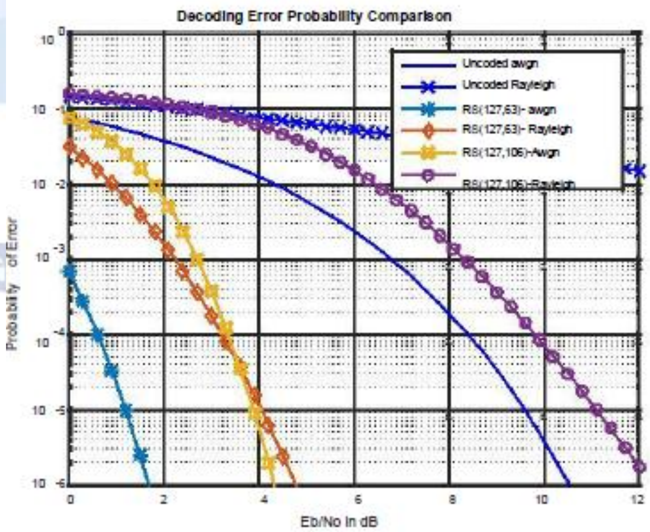

Fig. 2 Error performance of RS codes in different channels

Reed Solomon codes are optimal codes for need of robust communication, where the error does not degrade the performance of wireless channels. Hence in long distance channels with weak, noisy and distorted signal these codes are more efficient in correcting errors. They are suitable for burst error correction and as outer codes in concatenated coding system. WiMAX supports RS codes in forward error correction block for burst error correction[1].

Another category of codes called convolutional codes are used particularly to correct random errors in wireless channel.

These codes sum up the bits from the input data in a memory and use it to produce the output bits. The number of output bit generated per input bit gives the rate of convolutional codes. Compared to other coding system, the message bits are transferred in sequence instead of transferring as large blocks. The redundant bits are generated through modulo-2 convolutions in a convolutional encoder. Elias [15] developed the first algorithm for achieving zero error probability at a strictly positive rate. The algorithm introduced product codes and iterative decoding to the field. The idea of, data transmission at a rate maximum equal to channel capacity over a noisy channel and zero probability of error achievement at a positive rate firmly were departure from that period of coding research approaches. For reliable transmission at a rate approximately equal to channel capacity on noisy channel, search of suitable coding and decoding techniques are required.

The complexity of decoding can be reduced by breaking the computation into smaller manageable segments. This concatenation of large number of codes helps in decreasing the probability of error exponentially with the overall block length. Forney [16] established that the concatenation of a finite number of codes yields an error exponent that is inferior to that attainable with a single stage, but is nonzero at all rates below capacity. The maximum likelihood decoding (MLD) algorithm is used for convolutional codes. The upper bound and lower bound of the probability of error in decoding an optimal convolutional code in a memoryless channel is governed as a function of the constraint length of the code. It has been found that the performance of the convolutional codes is superior than the block codes of same length [17].

Convolutional codes are also decoded using a maximum a posteriori (MAP) decoder. The Viterbi ML decoding method minimizes the error probability of code word in convolutional codes. The method is based on the estimation of the complete sequence. It is also applicable to any trellis code. According to trellis the number of nodes in the trellis remains at 2K-1 (K is the number of flip flops used or shift register length or constraint length) though the number of input bits increases. For every path through the trellis, a 'metric' is calculated. Then the paths at every node that accurately balances the number of new paths created and large metric paths are discarded. This makes it possible to keepless number of paths containing maximum-likelihood choice.

The Viterbi is one of the most important decoding techniques for delivering coding gain for various types of channels. It can be hard-decision or soft decision decoding, depending upon the process of assigning values to the signals. In hard-decision decoding, the received signals are analysed by assigning the values of 1 or 0.A decision is taken about the condition of the transmitted signal, whether it is zero or one. These are provided as input to the Viterbi decoder. In soft-decision making, four regions are developed namely strong-one, strong-zero, weak-one and weak-zero. Also for the signals with less clear decision, intermediate values between zero and one are assigned. The soft decoding provides an improvement in coding gain up to 2-3 dB over the hard- decision Viterbi decoder for an AWGN channel.

The coding gain improvement in AWGN channel with BPSK or QPSK modulation is verified through software simulation and shown in the following figure (Fig.3). Though computational complexity is there in soft-decision decoding, it is preferred over hard-decision method due to the higher coding gain provided by the process.

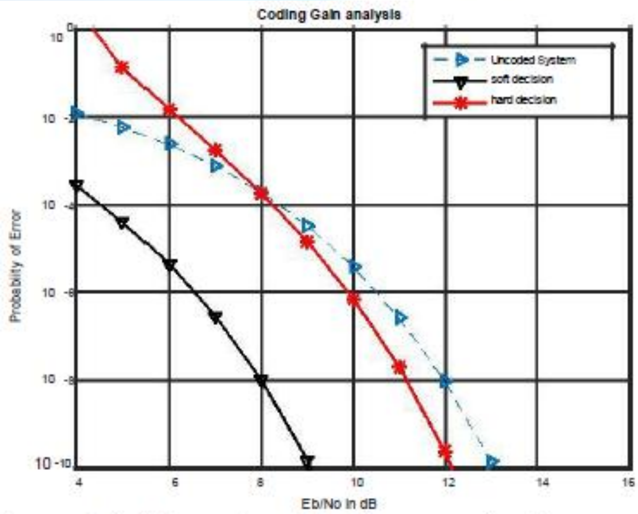

Fig.3 Soft decision and Hard decision comparison in AWGN channel

The major advantage of Viterbi decoding is its capability to easily operate on soft-decision data. The decoder receives the signal data in bits and follows the trellis. It searches for the most likely path of the trellis. The disadvantage of the Viterbi is that it cannot perform well in a burst channel. If the burst is longer than the largest trace back length, the decoder decodes wrongly resulting in a new burst of errors. One possible solution in those channels is considering the algorithm which distributes errors in more uniform manner to get low correlation between noise signals. At the same time, this adds delay in the processing. But this leads to significant delay in encoding, which may not be suitable for some specific applications [18].

The Bahl, Cocke, Jelinek and Raviv (BCJR) algorithm follows the MAP(maximum a posteriori probability) method. The a posteriori probabilities (APP) of the information and channel digits are obtained and used in the algorithm [19]. The decoding is based on the estimation of a single symbol considering the complete sequence of the received signals. The BCJR algorithm is used to process soft-input data based on a priori probability and providing soft-output data based on a posteriori probability. It calculates the reliability of a decision for a symbol by minimizing the symbol error probability.

Unlike the Viterbi, in another type called sequential decoding the computational complexity is kept independent of the code constraint length. This type of decoding can attain a satisfactory bit error probability, even for a large constraint length consideration in a convolution code. It involves sequential selection of most probable code words without searching the entire tree. The decoding complexity depends on the noise level [20]. The main advantage of the sequential decoding is its very little dependence on the channel characteristics. A quick change in the rate of decoding error probability is observed with increasing Eb/N0. It was Wozencraft who introduced this sequential decoding technique for decoding convolutional codes [21]. Fano developed this algorithm further with great improvement in decoding efficiency [22]. He offered a detail analysis of the decoding of digital signals after transmitting through a randomly disturbed channel. The search proc edure is like the hill-climbing type, in which the search of any set of alternatives is represented as a tree and the branches coming from different nodes of the same order are substantially different form one another. The sequential decoding technique is found to be more appropriate for low bit error rate (<10-5) requirements compared to Viterbi decoding. The disadvantage in comparison with Viterbi is the larger decoding delay. But it requires less storage than Viterbi and therefore it is more suitable for convolutional codes with large constraint length. Zigangirov[23] proposed a different sequential decoding procedure called stack algorithm. In this the decoder's memory is eliminated by increasing the number of steps for decoding. Unlike Fano decoder which stays in idle mode during low noise time intervals, stack never halts. The advantage of stack algorithm is the complete independence of the stack ordering and path extending portions of the algorithm. The speed increases as the coding rate is increased [24].

The major disadvantage of sequential decoding is its longer decoding delay compared to Viterbi decoder. The decoding delay of a sequential decoder for an n,k,m convolutional code is around n x B where B is the number of received branches that an input memory can adapt. The delay for a Viterbi decoder is just a small multiple (6 to 10) of n x m. The sequential decoding is also highly sensitive to channel parameters such as wrong estimate of channel SNR and incomplete compensation of phase noise. On the other hand, the Viterbi algorithm is robust for imperfect channel identification. The repetitive pipeline nature of the Viterbi algorithm makes it highly suitable for practical applications than sequential algorithm.

Turbo codes are a new class of convolutional codes with better band width and power efficiency. The performance in terms of BER are very close to Shannon's limit [25]. Also, turbo encoders are systematic convolution codes unlike the general convolution codes which are non-systematic. The turbo decoder consists of P pipelined elementary modules and rank p elementary module. The elementary modules use a complex algorithm which is a modified form of BCJR algorithm. [26] developed a simpler algorithm for Turbo-code decoding with complexity only twice that of the Viterbi algorithm but the performances are very close to the BCJR algorithm. The encoders and decoders are integrated in silicon providing good error correcting performance. As of now, Turbo are the most powerful error correcting codes at low SNR's. Also many of the binary block codes and convolutional codes are decoded using the Viterbi (MLD) or BCJR (MAP) soft-decision decoding algorithms. Hence, based on the end user requirement and acceptable probability of errors, the efficient algorithms are used for the given specifications.

The performance of block codes, convolutional codes and turbo codes are achieved by increasing the bandwidth of the transmitted signal. In some cases where bandwidth requirement is limited such as terrestrial microwave and some satellite channels, band width expansion is undesirable. The coding gain in such systems can be achieved by efficiently combining coding and modulation as a single entity. This is called as coded modulation which can be categorized based on the code structure as trellis coded modulation (TCM) and block coded modulation. In the TCM, at the same information rate, bandwidth and signal power, the minimum Euclidean distance between the coded modulation signals exceeds the minimum distance between the uncoded modulation signals. The objective of TCM is to generate a method that does the mapping of the coded bits into the signal symbols, so that the free distance between the coded signal sequences can be maximized. Ungerboeck[27] developed a method based on the principle of mapping by set partitioning.

Fountain codes are another class of codes used to develop an effective transmission system. These codes are developed to overcome the inefficiency of the random codes whose algorithms are complicated with longer quadratic running times for many applications. Luby developed the first class of fountain codes known as LT codes[28] with a fast decoder which is capable of retrieving the original symbol from any group of output symbols with high probability. To improve the performance further, another class of universal fountain codes known as Raptor codes [29] was developed which includes pre-coding of the input symbols before applying an LT code.

The LT codes are the first developed erasure correcting codes for any erasure probability. They offer successful communication over binary erasure channel (BEC). The LT code encodes a finite number of message symbols and generates unlimited number of output symbols. These output symbols are decoded by each receiver effectively. Due to its effective performance efficiency, the LT codes are successfully used in many applications. A transmitter with LT code uses only a single code for effective transmission of message through the broadcast networks. This is an ideal property of a fountain code. The main advantage of these codes is their ability to reconstruct the message even if the symbols get erased. There is a continuous effort taken by researchers to improve the performance of the LT codes and safeguard the data transmitted through internet by eliminating fading, noise and packet erasures. But the complexities of the codes get increased.

IV. Applications of Broadcasting

The conventional broadcasting systems like AM radio and FM radio systems are based on analog communication. But the advancements in digital technology software and hardware have led to the use of this technology in more information services. The application of broadcasting system is emerging with broader concept through the inclusion of data broadcasting, providing unprecedented variety and options.

The recent development in data broadcasting is the code transmission teletext systems. The problem in coded digital signals is the poor quality of the TV transmission paths. The objective of the teletext system is to improve this quality by transmitting digitised figures and characters overlapping with the TV signal. To overcome the error occurs in the system, highly capable code with large error correction ability is required. A decoder can be provided to the subscribers to make the decoder hardware compact and easy to handle. The teletext systems are under development in many countries such as Japan, Britain, Canada and United States. For Japanese teletext, a (272,190) shortened different set cyclic code is selected, which can be decoded by a simple majority logic circuit. The code is derived from simulation results based on bit-error-rate data collected from field tests. Another broadcasting service under development in Japan is the PCM sound via direct broadcasting satellite (DBSil). The errors due to multipath, impulse noise and waveform distortion can be neglected in comparison with bit errors due to rain attenuation, as that takes the major share of the errors. A (63,56) BCH code is used for error correction in this system.

V. Discussion

For moderate and high data rate systems, the convolutional codes with Viterbi decoding are found to be the most effective technique. The Viterbi technique is based on the assumption that there is no significant interference in the transmitted signal except the Gaussian noise. The main advantage of the Viterbi algorithm is its ability to process soft-decisions with ease as against the algebraic decoding algorithm which necessitates the use of hard-decisions. For processing very high data rates, concatenated Reed-Solomon code system is suitable, because it can give the same gain as that of Viterbi decoding, but with lesser complexity.

When large coding gains are required at high data rates, sequential decoding with hard decisions is found to be a suitable alternative to other algorithms. At moderate data rates, sequential decoding with soft decisions can be a better choice. In the case of mobile terminals, where the environment is operated in the existence of large Doppler offset and Doppler rate with multipath and fading errors, Reed-Solomon codes with soft-decision appears to be the best alternative to overcome the bursty type of the channel.

VI. Conclusion

The various error coding schemes and decoding methods have been analysed. The performance metrics of these codes depends on the communication environment like channel type, nature of the obstacles in an environment, rate of transmission, bit energy specification, Doppler frequency and the nature of error involved.

In actual mobile atmosphere the communication environment varies rapidly depending on the changes in vehicular speed. Under such conditions, the feedback data does not symbolize the current quickly changing environment based on the changing speed. This also leads to unfavourable impact in the effective transmission data rate and the channel becomes unreliable. The reliable performance of the reactive FEC schemes is affected in the case of wireless communication where the end nodes are moving at changing vehicular speeds. Therefore, more effective FEC schemes like, pro-reactive error control codes are required to be developed to face the current challenges.

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