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Implementation of Turbo Codec for Differentiated QOS Applications

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Abstract: In this study, an adaptable Turbo codec has been designed to meet the specifications of different Quality of service (QOS) of 3G applications. In the third generation networks, a wide variety of services are offered and the different services require reconfiguration of the existing Turbo codec. There are different stages involved in the design of turbo codes. The design varies according to the requirements of quality of service. Certain applications may require less delay, whereas some other application may need less loss in data. Turbo codes can be made to satisfy the different QOS requirements by proper analysis and design. In this study, a complete analysis has been made on certain QOS requirements and the impact of various components of turbo codes on the QOS. Based on the analysis alongwith the simulation, the design of turbo code is implemented to achieve the desired Quality of service for various transmission.

Key words: Turbo codes, QOS parameters, bit error rate, interleaver, generator polynomial

INTRODUCTION

The Third Generation networks and services present opportunities to offer various data applications and service that meet end-to-end quality of service. The concept of Quality of Service (QOS) arises due to the fact that wireless communications require guarantees for effective and comprehensible transmission of audio and text data content. QOS can be specified as a set of parameters that satisfy the desired qualitative and quantitative requirements. The user's auditory and visual perceptions define the acceptable parameter values and the acceptable QOS.

The main features of Universal Mobile Telecommunication Services (UMTS) are Variable Quality of service (BER between 10^{-3} and 10^{-6}, delays between 30 and 300 ms).

There are certain applications in real systems operating at relatively low BER's such as data transmission in satellite and space communications. Some of the applications which operate at high BER are voice transmission in mobile communications and military communications.

The main aim of this study is to define the QOS parameters and to find the related parameters used in the design of turbo codes. In section I the basics of turbo codec is dealt, section II deals with the QOS parameters. The various components used in turbo codes and their impact on the QOS is dealt in Section III. Finally conclusion and future work is discussed in section IV.

TURBO CODES

Error control codes have become a vital part of modern digital networks, enabling reliable transmission to be achieved over noisy and fading channels. Turbo codes have been widely considered to be the most powerful error control code of practical importance. Turbo codes emerged in 1993[1-3] and have since become a popular area of communications research. The important characteristics of turbo codes are the small BER achieved even at low Signal to Noise Ratio (SNR) and the flattening of the error rate curve (i.e.) the error floor at moderate and high values of SNR. The performance of turbo codes is due to the randomness created by the interleaver and the iterative decoding. The different blocks of turbo codes is explained in detail in the following sections.

Turbo encoder: The basic structure of turbo code encoder is shown in Fig1. It consists of two binary Recursive Systematic Convolutional (RSC) encoders with small constraint lengths usually set between 3 and 5 which are concatenated in a parallel fashion by using a Turbo code interleaver and a puncturing and multiplexing device. The

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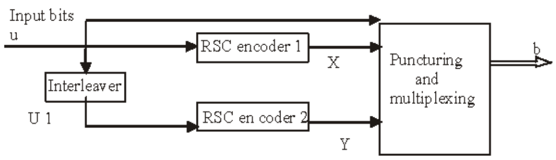

Fig. 1: Structure of turbo encoder

binary input sequence u which has finite duration is fed into the first RSC encoder yielding the redundancy sequence X with the same finite duration as u. Sequence u1 which is an interleaved replica of u is put into RSC encoder 2. The output of RSC encoder yields redundancy sequence Y. Both the redundancy sequences X and Y as well as u are punctured and multiplexed to form output sequence b. In Fig. 1, the constraint length of the encoder is K and memory is m = K-1. The input to the encoder at time k is a bit u_k and the corresponding binary output

$$X_k = \sum_{i=0}^{v} g_{1i} u_{k-i} \tag{1}$$

$g_{1i} = 0,1$

$$Y_k = \sum_{i=0}^{v} g_{2i} u_{k-i} \tag{2}$$

$g_{2i} = 0,1$

where g_{1i} and g_{2i} are the two-encoder generators.

Turbo decoder: There are two main algorithms to decode the data. One of the algorithm is called Soft Output Viterbi Algorithm (SOVA) and the other one is Maximum A Posteriori Algorithm (MAP). The first algorithm finds the most probable output data sequence. The trellis diagram is drawn and the path with the least Hamming distance is found. The MAP algorithm finds the marginal probability that the received bit was a 1 or a 0. Since the bit could occur in many different codewords, the sum of all the probabilities are considered. The decision is made by using the likelihood ratio of the marginal distributions from 1 and 0. The calculation is structured by using the trellis diagram. The decoder examines the states and computes the a posteriori probabilities associated with the state transitions. The log likelihood ratio of the estimated bit is the division of the probability that the transmitted bit u_k = 1 to the probability that the bit was u_k = 0. The MAP algorithm is preferred as it minimises the bit error probability. The iterative decoding algorithm is done with the MAP algorithm. It consists of two component decoders serially concatenated via an interleaver. The first MAP decoder takes as input the received information sequence and the received parity sequence generated by the first encoder. The decoder then produces a soft output, which is interleaved and used to produce an improved estimate of the a priori probabilities of the information sequence for the second decoder. The other two inputs to the second MAP decoder are the interleaved received information sequence and the received parity sequence produced by the second encoder. The second MAP decoder also produces a soft output, which is used to improve the estimate of the apriori probabilities for the information sequence at the input of the first MAP decoder. After a certain number of iterations, the soft outputs of both MAP decoders stop to produce further performance improvements. Then the last stage of decoding makes a hard decision after deinterleaving.

Turbo interleaver: Interleaving is a process of rearranging the ordering of a data sequence in a one to one deterministic format. The inverse of this process is called deinterleaving which restores the received sequence to its original order. Interleaving is a practical technique to enhance the error correcting capability of coding. In turbo coding, interleaving is used before the information data is encoded by the second component encoder. The basic role of an interleaver is to construct a long block code from small memory convolutional codes, as long codes can approach the Shannon capacity limit. Secondly, it spreads out burst errors[4,5]. The interleaver provides scrambled information data to the second component encoder and decorrelates inputs to the two component decoders so that an iterative suboptimum-decoding algorithm based on uncorrelated information exchange between the two component decoders can be applied. The final role of the interleaver is to break low weight input sequences and hence increase the code free Hamming distance or reduce the number of codewords with small distances in the code distance spectrum.The size and structure of interleavers play a major role in the performance of turbo codes. There are a number of interleavers, which can be implemented.

The random interleaver[3] uses a fixed random permutation and maps the input sequence according to the permutation order.

The block interleaver[3] is the most commonly used interleaver in communication system. It writes in column wise from top to bottom and left to right and reads out row wise from left to right and top to bottom.

Diagonal interleaver writes in column wise from top to bottom and left to right and reads out diagonally from left to right and top to bottom.

The permutation P of the circular-shifting interleaver is defined by

$$P(i) = (ai+s) \bmod L \qquad (3)$$

QOS PARAMETERS

QOS refers to the capability of a network to provide better service to selected network traffic over various network technologies. In this study, we have considered only three parameters and in the next section, we analyse the design and configuration that has to be made in turbo codes to maintain these QOS parameters.
QOS provisions over wireless networks are

- Latency
- Loss
- Data Rate

Latency: Latency is the amount of time it takes a packet to reach the receiving endpoint after being transmitted from the sending endpoint. This time period is termed the end-to-end delay. An upper bound on the delay Dmax denotes the maximum time any packet of a stream will need to be transferred. For real time applications the maximum allowed delay is restricted. The allowed delay of packets determines the priority of transmission. If the transmission is audio, then the delay has to be less, whereas in the text transmission the delay can be an affordable factor.

Loss: Loss is a comparable measure of packets faithfully transmitted and received to the total number that were transmitted. It is expressed as the percentage of packets that were dropped. The bit error rate is an important parameter of QOS. Relatively high BER can be allowed for speech communication (10^{-2}). Low BER should be achieved for Data communication (10^{-6}). The data loss or the bit error rate varies due to the varying channel conditions especially in the mobile cellular environments where the channel is subjected to different environments.

Data rate: Data rate is the amount of information that can be transferred per unit time. It is referred to as bandwidth for the networks. There are various requirements of the QOS where the data rate has to be changed to efficiently utilize the bandwidth.

CONFIGURATION OF TURBO CODES FOR DIFFERENTIATED QOS

The performance of turbo codes is governed by lot of factors. By proper[3] design and configuration, the

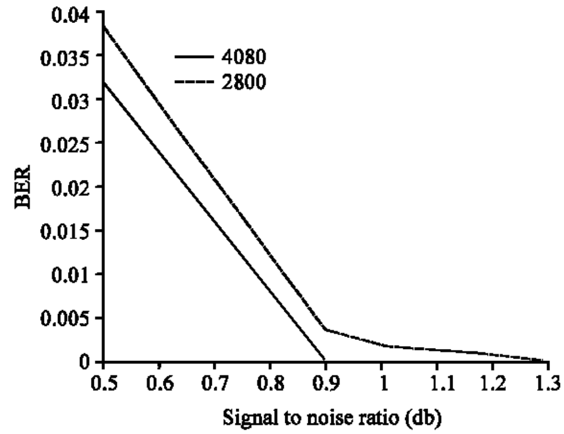

Fig. 2: Performance analysis of turbo codes for varying frame sizes

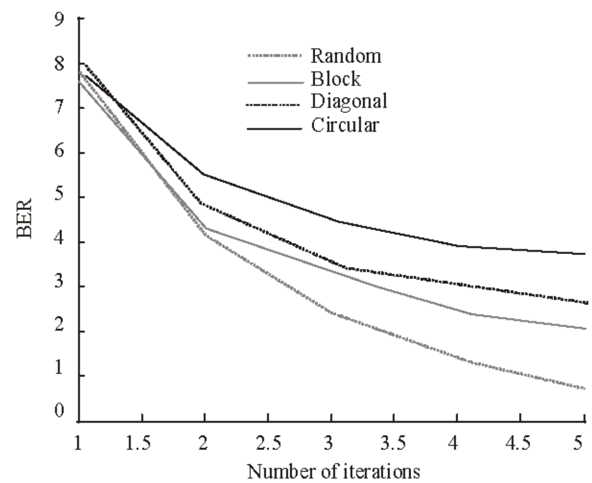

Fig. 3: Performance analysis of turbo codes for varying interleaver structures

performance can be largely improved. The different factors are discussed in the following section.

Size and structure of interleaver: The proper design of interleaver plays a major role in the performance of turbo codes. As the interleaver enables the information exchange between the two component decoders, increasing the interleaver size has the effect of randomizing the information sequence at the input of the second decoder. Consequently, the inputs to the two component decoders become less correlated, with respect to noise, improving the decoding performance.The design criterion at high SNR or low BER is to maximize the code effective free distance. At low SNRs or high BER, the performance of the code is determined by its distance spectrum. The interleaver structure and size affect the turbo code error performance considerably. At low SNRs the interleaver size is the only important factor as it can increase the interleaver gain and enhance the code performance. The effects induced by changing the interleaver structure at low SNR region are not significant. The interleaver structure affects the mapping of low weight input sequences to the interleaver output and hence the first several distance spectral lines of the turbo code distance spectrum. It plays an important role in the region of high SNR's. At high SNRs both the structure and size play an important role in maximizing the code effective free distance, thus enhancing the performance of the turbo code.The performance analysis of turbo codes for varying interleaver sizes and structures is shown in Fig. 2 and 3.

Latency is given by

$$t_d = \frac{K_f}{R_b} N_i \tag{4}$$

where R_b is the bit rate, K_f is the frame size and N_i is the number of decoding stages. From the formula we are able to conclude that latency is directly proportional to the frame size. Larger the frame size, larger is the latency or the delay. For applications where delay is a major constraint, then the frame size should be reduced to get less delay or fast transmission. The above formula has been applied to two different text frames whose simulation results are shown in Fig. 2. The bit rate considered is 128 kbps and the number of iterations is 5. We are able to see that for larger frame size of 4080 bits, the latency is 159.4 ms and for smaller frame size of 2800 bits, the latency is 109.4 ms.

Turbo codes have been shown to exhibit remarkable performance when the frame size is large. The size of the interleaver is same as the size of frame. However, the latency in a turbo-coded system is directly proportional to the frame size. Thus, there is a direct tradeoff between latency and Bit Error Rate (BER) based on the choice of interleaver size. This tradeoff can be exploited in multimedia communication systems by using different interleaver sizes to achieve different QOS.

Choice and design of constituent codes: The Turbo encoder consisting of two constituent Systematic Recursive convolutional encoders is linked by an interleaver. The first encoder operates directly on the original data sequence and the second encoder operates on the interleaved version of the same data. This construction can be generalized to any number of constituent codes. The codeword of the turbo encoder

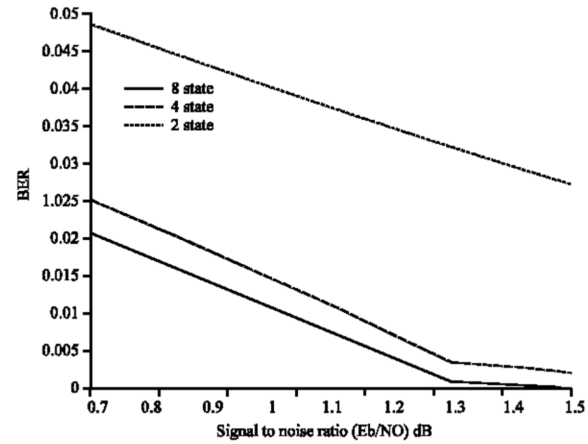

Fig.4: Performance analysis of turbo codes for varying constraint lengths

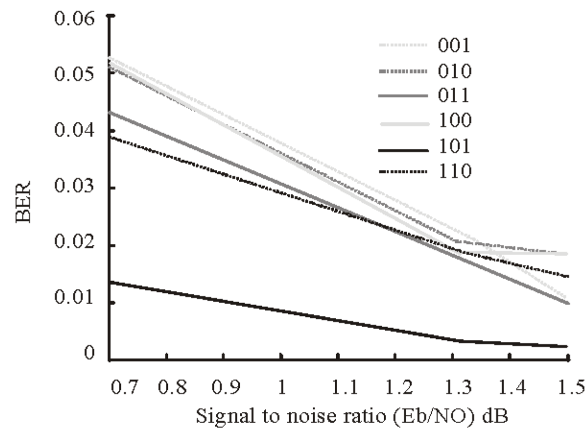

Fig. 5: Performance analysis of turbo codes for varying generator polynomials

consists of the information bits followed by parity check bits of both encoders. The performance of turbo codes with same interleaver size is categorized in two-ways.

Variable constraint length: Increasing the memory order of component codes results in an increase in the effective free distance and hence the performance.Increasing the memory order increases the complexity of the hardware and hence the delay increases. For certain applications where the interleaver size is less, increasing the memory order does not increase the performance. In this study an audio file has been transmitted with different constraint lengths and the simulation results are shown in Fig. 4.

Varying generator polynomial: At high SNR's, the interleaver size has to be large. Hence the design criterion at high SNR would be to maximize the code effective free distance. Therefore the design of constituent codes should lead to maximizing the free distance of the component recursive convolutional codes. Many researchers have suggested various methods by which the free distance can be maximized. Benedetto and Montorsi[6] and Divsalar, Pollaro[7] have suggested that the feedback polynomial used in the RSC component encoder should be a primitive polynomial.

The generator matrix of a rate ½ component RSC code can be represented as

$$G = \left[1, \frac{g_2(D)}{g_1(D)} \right] \tag{5}$$

where $g_2(D)$ and $g_1(D)$ are feed forward and feedback polynomial.

The design algorithm[6] is applied to select the generator polynomials g_1 and g_2. The algorithm is as follows

- Choose as g_2 (D) a primitive polynomial of degree ν.
- Choose $g_1(D)$ to be a monic polynomial of degree ν.
- Both $g_1(D)$ and $g_2(D)$ should be relatively prime.
- For different values of $g_2(D)$, choose the value yielding the lowest error probability for the desired interleaver length.

The design algorithm has been applied to an audio file. The polynomial $g_1(D)$ is fixed as 111 which is a primitive polynomial. The choice of $g_2(D)$ is shown in the simulations in Fig.5 where the graph is plotted for various noise levels and bit error probability. From the results, it is obvious that the lowest error probability is given by $g_2(D) = 101$.

Choice of decoding algorithm: There are two main algorithms used in the decoding of turbo codes. One is the MAP algorithm and the other is the SOVA. The first algorithm finds the most probable bit that was transmitted and the second algorithm finds the most probable path or sequence that was transmitted. Both the algorithms can be used as an iterative algorithm. SOVA is less complex compared to MAP but the performance of MAP is far better compared to SOVA. Each algorithm is represented by the number of computation operations for an (n,k) convolutional code of memory order v. For an iterative decoder, the complexity is proportional to the number of stages. Hence where the data loss is not a major criterion, SOVA can be employed and for applications where the performance has to be maintained with affordable delay, MAP algorithm can be employed.

CONCLUSION AND DISCUSSION

Turbo codes has been designed and implemented with various parameters which can be adapted to our requirements. A complete analysis has been done on the design of turbo codes for various QOS requirements. A tradeoff has to be made between the delay and performance on the design of turbo codes. Based on the QOS requirements, the choice of turbo codes can be made. In this study, we come to a conclusion of the various factors of turbo codes suitable for the QOS parameters Latency and Loss. The analysis has been applied to an audio and a text data. The discussion is as follows.

In audio transmission, the delay is an important factor but data loss can be tolerable. Hence the design of turbo codes has to be made as per the condition that delay is less and loss is more. The choice of interleaver can be made such that the frame size is small. The constraint length has to be less and the code design can be made as per the design[6]. SOVA decoder can be used since the performance is not a major factor.

In text transmission, the delay can be affordable. Hence the interleaver size can be large and the constraint length can also be more. MAP algorithm has to be employed since the performance has to be maintained.

The future work can be to optimize the parameters required using any optimization techniques.

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