Quality of Service (QoS) for a communication system with applying the quantization of Turbo Code

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Abstract

Ensuring the quality of service (QoS) is a very important topic in wireless communication. So we investigate the effect which derives from quantization upon turbo code with powerful error control technique. By choosing quantization bit to reach the QoS that we want.

Keywords: QoS, turbo code, quantization bit

Introduction

In recent ten years, there are many researchers invest a lot of time in developing the wireless communication technology, and they have obtained many research achievements. Now, these technologies have been widely employed to advance the industry science in order to satisfy requirements for third generation wireless communication system (3G). In 3G, many kinds of messages are transmitted between two communication terminals, for example, voice, audio and text, etc. Each once needs different QoS, so resources of communication system must been managed for ensuring the quality of various traffic. CPU operating time is one kind among these system resources [1]. In this paper, we wish to reach the purpose by researching the effect of quantization process upon channel code.

In the wireless communication system, all real-life transmission channels are noisy, so the received signal is surely distorted due to interference [2]. That would make the detector to do the wrong decision, and it would result in the reduction of system performance. The type of error could be categorized as random error and burst error. According to previous research, random error and burst error could be eliminated by using a convolutional coding scheme and interleaver coding scheme, respectively [3]. Error control coding scheme could be employed to correct the error data. It helps the communication system to obtain a reliable reproduction of data and makes the performance having a great improvement under a heavy interference environment.

Channel coding refers to the class of signal transformations designed to improve communication performance by enabling the transmitted signals to better withstand the effects of various channel impairments, such as noise, interference, and fading [3]. It endows a communication system the capability of correcting error bits to improve the performance. Since the channel coding technique was developed, the researchers and engineers have proposed many coding arts. For example, Hamming code, convolutional code and turbo code etc [3]. In these, turbo code is the best coding scheme for it could quite close to the Shannon limit [4]. It was proposed by Berrou in 1993. Berrou combines the concatenated code with interleaved code to propose the turbo encoding. In this coding scheme, two recursive systematic convolutional (RSC) codes are connected in parallel, and they are separated by a random interleaver. In past, a convolutional code with better performance means that the structure of its encoder needs more registers, and the operating process of its decoding is more complex. Using concatenated code could achieve the purpose which has good performance without increasing the decoding complexity. Besides, the primary reason for using a long interleaver in turbo coding is to generate a concatenated code with large block length which leads to a large coding gain. Above two reasons let the codeword have a powerful capability to correct error bits. Berrou employed tow decoders to construct an iterative decoding scheme, and used the MAP (maximum a posteriori) decoding method to be the algorithm [4]. During the decoding process, the information is exchanged between two decoders, and the error probability decreases gradually with the increase of iterative times. Then, a low bit error rate could be achieved at a signal-to-noise ratio close to the Shannon capacity limit. However, the amount and the complex degree of calculation in MAP algorithm are very high. Therefore, those later researchers proposed new algorithms which have lower complexity in calculating. For example, log-MAP, MAX-log-MAP [5] and SOVA (Soft output Viterbi algorithm) [6] etc.

These algorithms are theoretical. They are proposed under the assumption that the input data of decoder is analog, but the quantizer is needed in receiver to produce a discrete representation of the waveform in practice. Because the quantization process will cause the difference between input and output data, the decoding performance is not the theoretical value. Besides, an adapted quantization bits will decrease the input data of decoding and raise the decoding rate. Recently some work was done to evaluate the effects of quantization data on the MAP, log-MAP, and the SOVA algorithms [7]-[9]. These researches use AWGN (additive white Gaussian noise) channel model to be the simulation environment, but the channel model of real wireless communication is more complex. In this paper, it uses the Rayleigh fading channel to be the system environment and to discuss the effects of quantization data on decoding performance.

The configurations of turbo encoder have been established in 3rd Generation Partnership Project (3GPP) [10], so we employ the turbo encoder of 3GPP standards to be the scheme in simulation. Besides, we apply the Bi-SOVA (bi-directional soft output Viterbi) algorithm [10] to turbo decoder and observe the effect of quantization on the system transmission efficiency. The remainder of the paper is organized as follow: In section II, the configuration and principle of turbo code is introduced. Then, it describes the whole communication system and the procedure of signal process. In section III, the quantization configuration is introduced. It illustrates that how is the received signals quantized and the effect of quantization on information. In section IV various simulation results are presented. Finally, section V concludes.

The system model

The communication system model is showed as Fig.1. After the process of wireless communication, the transmitted waveform is usually different from received waveform, and it would cause errors in the received data. For this reason, we use turbo encoder to provide the original data (dk) with a powerful error correction capability. These encoding data would be modulated and transmitted to the receiver. The radio propagation environment affects the performance of wireless communication systems. The phenomenon is affected by prominent terrain contours between the transmitter and receiver. Besides, in the delivered process of wireless communication, the transmitted signal may be reflected and refracted by obstructers. The resultant received signal because of the multipath propagation effect is the sum of signals through an infinite number of paths. This multipath propagation phenomenon gives the received signal a fading effect. The fading effect is called Rayleigh fading if there are multiple reflective paths that are large in number, and if there is no line-of-sight signal component. The envelope of such a received signal is described by Rayleigh statistics.

The transmitted signal would also be interfered by the extrinsic noise so the system channel model is composed of additive white Gaussian channel (nk) and fading channel. In the receiver, the received signal would be demodulated and quantized to turn the analog signal onto digital information. The digital information would be decoded and then we could get the expected information (dk').

Fig. 2 shows a classical block diagram of a 1/3 rate turbo encoder. It is composed of two identical RSC encoders and a random interleaver. An encoder is parallel with another, and connects to each other by the interleaver.

The generator matrix of a 1/2 rate RSC encoder can be represented as

$$G(D) = \begin{bmatrix} 1 & \dfrac{g(D)}{f(D)} \end{bmatrix} \tag{1}$$

where g(D) and f(D) are a feedforward and feedback polynomial respectively, and the symbol D is called the delay operator. If the input data stream is a binary sequence, given by C, the codeword V can be derived by the encoding operation which is represented in a matrix form as

$$V = C \cdot G \tag{2}$$

where all operations are modulo 2. In the turbo encoder, C would be directly delivered to be V1 without doing any operation and be encoded by RSC1 to produce the parity check sequence, denoted by V2. The convolutional code has powerful capability to correct random errors, but it is not for burst errors. Turbo encoder uses a interleaver to turn the burst errors into random errors, and RSC2 encodes the interleaved sequence, denoted by C', to produce second parity check sequence, denoted by V3.

Fig. 3 is the structure of turbo decoder. In the process of wireless communication, radio and quantization would cause some differences between codeword and input data of decoder so we use V1', V2' and V3' to present transmitted codeword. Decoding is a process that it employs the maximum a posteriori criterion to guess the transmitted codeword. In another word, decoding algorithms are to employ the input data of decoder to calculate the probability that the transmitted information is '0' or '1' in each time sequence. If the happened probability of '0' is grater than '1', we do the decision that transmitted bit is '0'. On the contrary, we do the decision of '1'. According to the MAP criterion, we obtain a useful metric called the log-likelihood ratio (LLR). It is defined as

$$L(d|V') = \log\left[\frac{P(d=+1|V')}{P(d=-1|V')}\right] \tag{3}$$

Let the binary logical elements 1 and 0 be represented electronically by voltages +1 and -1, respectively. The symbol d is used to represent the transmitted data bit. According to the Bayes' theory, this equation of LLR can be decomposed as follow:

$$L(d|V') = \log\left[\frac{P(V'|d=+1)P(d=+1)}{P(V'|d=-1)P(d=-1)}\right]$$
$$= \log\left[\frac{P(V'|d=+1)}{P(V'|d=-1)}\right] + \log\left[\frac{P(d=+1)}{P(d=-1)}\right] \quad (4)$$

To simplify the notation, equation (4) is rewritten as

$$L'(d') = L_C(V') + L(d) \quad (5)$$

Where the notation LC emphasizes that the LLR term is the result of a channel measurement made at the receiver, and L(d) is the priori LLR of the data bit d. For a systematic code, it can be show [3] that the LLR out of the decoder is equal to

$$L(d') = L'(d') + L_e(d') \quad (6)$$

Where L'(d') is the LLR of a data bit out of the demodulator (input to the decoder), and Le(d'), called the extrinsic LLR, represents extra knowledge that is gleaned from the decoding process. The output sequence of a systematic decoder is made up of values representing data bits and parity bits. From equation (5) and (6), the output LLR of the decoder is now written as

$$L(d') = L_C(V') + L(d) + L_e(d') \quad (7)$$

Decoding device uses this equation to calculate the extrinsic LLR and passes the information to another decoder via interleaver or de-interleaver. After several times of iterative operation, turbo decoder could get the value of LLR (L2) and do the binary decision to get the expected information.

Quantization configuration

Amplitude quantization is a procedure which transforms a given signal amplitude x(n) at time n into an amplitude y(n) taken from a finite set of possible amplitudes. As shown in Fig. 4, the signal amplitude x is specified by index k if it falls into the interval

$$\Psi_k : \{x_k < x \le x_{k+1}\}; \quad k = 1,2...,L \quad (8)$$

The L-ary mumber k is transmitted to the receiver, typically in binary format. Let L=2R; then a bit rate of

$$R = \log_2 L \quad (9)$$

is needed to inform the receiver about that index. At the receiver, the index k is transformed into an amplitude yk that represents all amplitudes of the interval Ψk. The amplitude yk are called the representation levels or the reconstruction values, and the amplitudes xk are called decision levels or decision thresholds. The number of intervals equals the finite number L. The set of possible outputs is described by

$$y \in \{y_1, y_2, ..., y_L\} \quad (10)$$

Unless otherwise mentioned, we assume error-free transmission. We thus have an output

$$y = y_k \quad \text{if} \quad x \in \Psi_k \quad (11)$$

For a close approximation to the input, one requires that yk □ Ψk . The actual mapping

$$y = Q(x) \quad (12)$$

is the quantizer characteristic, a staircase function by definition. Inherent in the quantization process is an error between input x and output y. The error is called quantization error, and we shall define it as the difference between x and y:

$$q = x - y = x - Q(x) \quad (13)$$

Because of the existence of quantization error, the decoding performances between quantized information and quantized information are different. In this paper, we observe the BER (bit error rate) on different quantization condition and discuss the effects of data on decoding performance [11].

Simulations and results

Fig. 5 is the standard of turbo encoder for 3GPP, and the interleaver of length N=1024 is used. In this paper, we use it to be the encoding device and employ Bi-SOVA to be the decoding algorithm. In the simulation environment with Rayleigh fading channel, we observe the effects of quantized data on decoding performance.

In Fig. 2, the received soft data is turned into the discrete data by a uniform quantizer. The uniform quantizer divides the quantization range, [-A, A], uniformly into L=2R intervals. The quantized value is chosen to be at the center of its corresponding quantization interval. As show in Fig. 6, we use a statistic method to find the value which is the ninety percent magnitude of received signal and use it to be the value of A. In the simulation, A and R are varied to study the effect of quantization range and number of quantization bits on the turbo code performance.

As show in Fig. 7, it is the effects of different iterative decoding times on decoding performance when the simulation channel is AWGN model. In the simulation, the input data of decoder is unquantized. These curves of decoding performance will tend towards a limit value with the increase of iterations. For increasing the decoding efficiency, we find a curve which has good performance and less iteration number. Then, the number of iterative times can be used in after simulations. From the simulation result of Fig. 7, we consider that four times is the most adapted iteration for turbo decoder under AWGN channel model. Beside, Fig. 8 shows the simulation result which is similar to Fig. 7 but it is under the Rayleigh fading channel model. By the same reason of above representation, we use three times iteration in turbo decoder under Rayleigh fading channel. In the next experiences, we discuss the influence of quantization data on decoding performance.

Fig. 9 and Fig. 10 are performance curves of different quantization bits under AWGN channel and Rayleigh fading channel respectively. These curves are gradually close to the theoretical value with the increase of quantization bit. From them, we know that the performance could almost coincide with the theoretical curve when the quantizer uses five bits to describe the input data under AWGN channel and seven bits under Rayleigh fading channel.

Conclusion

In this paper, we wish to manage the communication resource, CUP operating time, by choosing the adapted quantization bit, and we can use it to ensure the QOS. For example, we need seven bit to describe voice information in higher noise situation (SNR=13). When the noise is reduced (SNR=14), we could use four bits to present input data of decoder. At this time, the CUP operating time is decreased with the reduction of quantization bit. Besides, the decoding devices with Bi-SOVA algorithm are used in the paper. Because the decoding algorithm is low complex [12], it also decreases the CPU operating time. From above represent, we know that using adapted quantization bit and low complex algorithm, Bi-SOVA, could reach the requirement of QOS.

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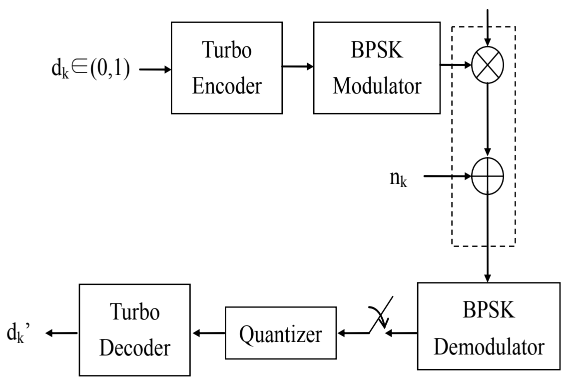

Fig. 1 The communication system model

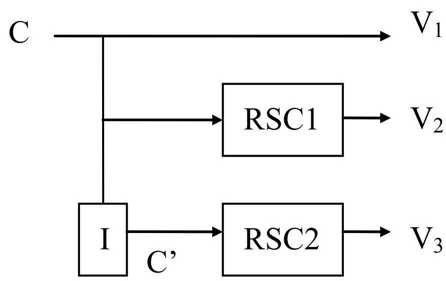

Fig. 2 The turbo code encoder

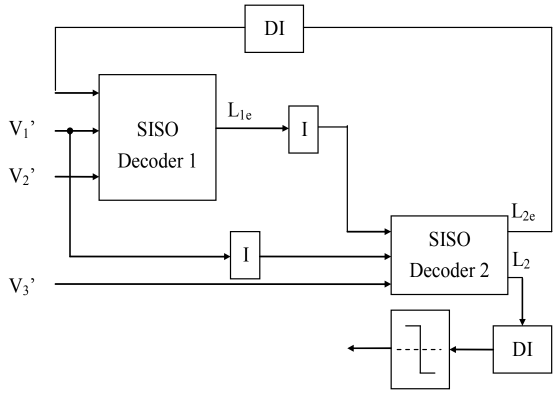

Fig. 3 Turbo code decoder

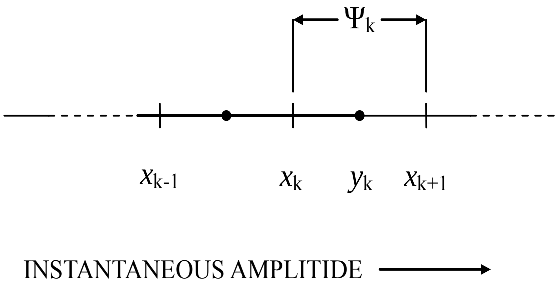

Fig4. Quantization of received signal

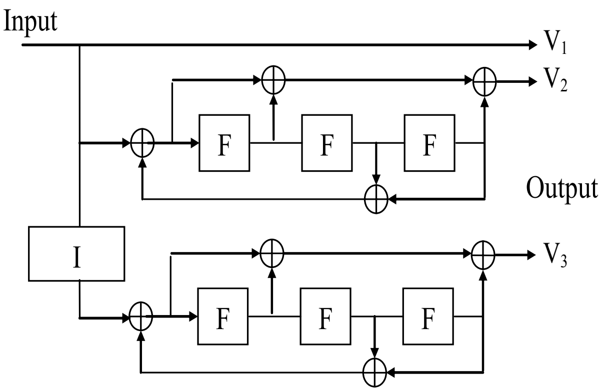

Fig. 5 The turbo encoder constructor for 3GPP

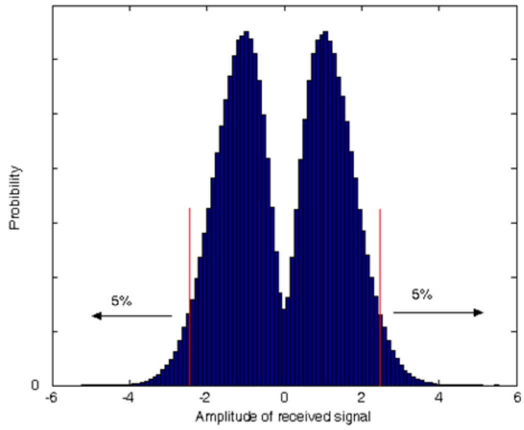

Fig. 6 The chosen quantization range

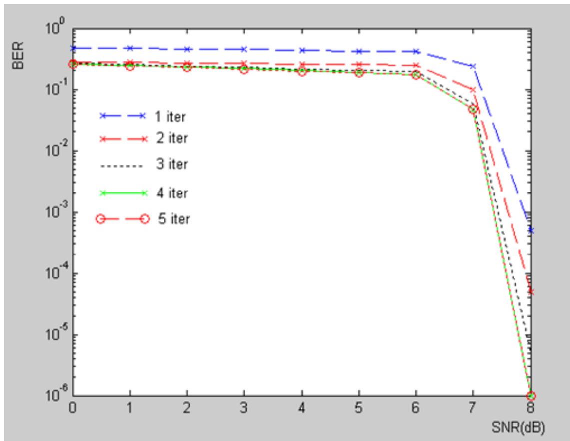

Fig. 7 Effect of iterations on turbo code performance using the Bi-SOVA under AWGN channel

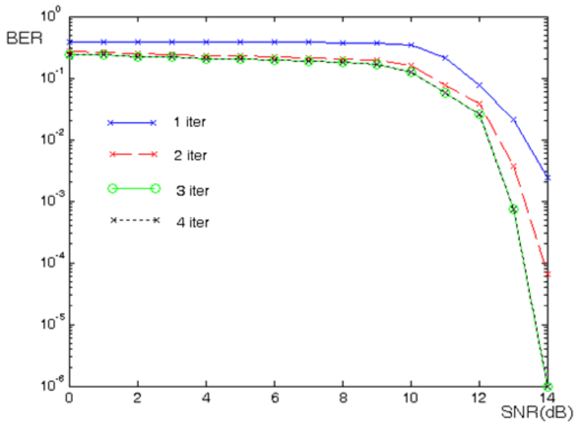

Fig. 8 Effect of iterations on turbo code performance using the Bi-SOVA under Rayleigh fading channel

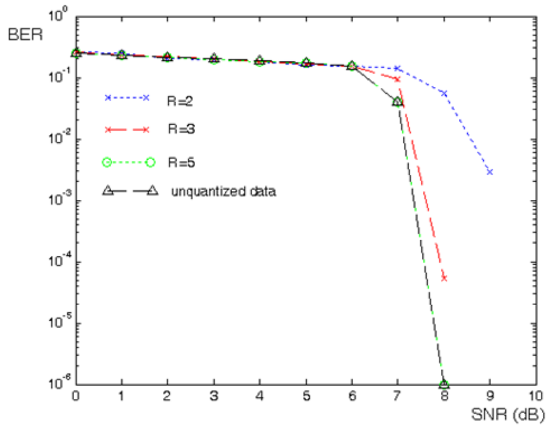

Fig. 9　Effect of quantization bits on turbo code performance under an AWGN channel

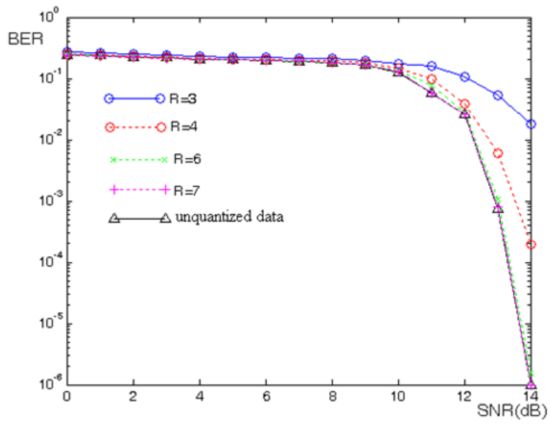

Fig. 10　Effect of quantization bits on turbo code performance under Rayleigh fading channel