COQRC: A Rateless Video Transmission Solution

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***Abstract*—The explosively growing wireless video traffic demands high video quality and transmission efficiency. A new scheme for video transmission over channel with noise is proposed in this paper. Like standard video transmission, the scheme includes entropy coding followed by powerful channel coding. Unlike conventional schemes, we use a rateless coded modulation scheme instead of adaptive modulation coding (AMC) in traditional digital video transmitting to solve the dilemma of inaccurate channel estimation. The proposed rateless scheme does not need the explicit channel condition, it matches channel condition implicitly. We optimize the proposed system over GF(2). Performance comparisons are made with respect to the state-of-the-art video coding and channel coding scheme used in Long Term Evolution (LTE) mobile network. Experiments show that the proposed system outperforms LTE by 2-3dB.**

***Index Terms*—Wireless Video Communication, Rateless Coding.**

I. Introduction

According to Cisco VNI Forecast[1], mobile video traffic accounted for 60 percent in 2016. Mobile data traffic caused by video will account for over 3/4(78 percent) of the worlds mobile data traffic by 2021, increasing 9-fold than 2016. Conventional digital video transmission over noisy channels is designed based on Shannon's source-channel separation theorem. Video traffic is implemented by a linear transformation like Discrete Cosine Transform, followed by quantization. The quantized bits are entropy coded and transmitted after channel coding. Due to the waterfall behavior of the channel coding, fluctuation in the transmission channel can cause dramatic degradation of the decoding Bit Error Rate (BER), causing a large number of packets retransmission and bandwidth-wasting [2].

In this paper, we consider the utilization of the Rateless Coded Modulation scheme put forward in [3] to deal with the problem of wireless video transmission. The proposed technique, referred to as Channel Optimized Quantization with Rateless Coding(COQRC), maps the entropy coded bit sequence directly into modulation symbols instead of conventional channel coding and modulation. Unlike conventional scheme based on concatenating sources coding and channel coding which are optimized independently, the proposed scheme optimizes these two components jointly. Research has shown that the quantization parameters as well as channel coding schemes are essential for maintaining a good transmitting performance. In general, there are a variety of rateless coding schemes aiming at adapting noisy channel conditions without feedback [3][4][5]. The transmitter progressively produces symbols with fine-grained bit energy allocation. A survey of the rateless coding schemes is provided in section II.

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If the transmission channel is Additive White Gaussian Noise(AWGN), COQRC achieves near optimal performance given by Shannon's theorem with the restriction of finite block length and affordable decoding complexity. COQRC has a better PSNR performance than the conventional source and channel coding.

The rest of the paper is organized as follows. Section II introduces backgrounds and related work. Section III includes the framework of the proposed system. We choose Rateless Coded Modulation (RCM) codes as the channel code of the COQRC scheme in view of their rate flexibility. Section IV focuses on the optimization for the specific COQRC application. Section V introduces the de-convolution belief propagation algorithm used by the receiver. In section VI, we evaluate the performance of COQRC and compare with the state-of-the-art video transmission scheme presently used by LTE. Finally, we conclude the paper in section VII.

II. Background and Related Work

Our work is related to the work on the rate-adatption video coding and rateless coding.

Rate Adaption: Wireless channels are time-varying because of the Multipath Effect and Doppler Effect of the movement. Choosing the lowest rate of channel coding and modulation guarantees the transmission stability but reduces efficiency.

With Adaptive Modulation Coding (AMC), transmitter adjusts coding rate and modulation scheme according to channel conditions. However, in reality, this technology is not very efficient because of two reasons: first, it is difficult to ensure the accuracy of channel estimation since estimations base on a finite number of reference signals. Second, there is staircase effect in AMC. The number of encoding and modulation rate is limited either it is a local area network (LAN) 802.11 or mobile communication network LTE, thus the throughput is not smooth. Even if with accurate channel estimation, AMC can only achieve discrete staircase-like rate adjustment (throughput curve like a staircase). The discontinuity between staircases lowers the efficiency of communication.

Hybrid Automatic Repeat Request (HARQ), first introduced in[6], encodes the bitstream using a low-rate code and transmits subpackets in turn using an automatic repeat request (ARQ) algorithm, until the receiver decodes the bitstream correctly.

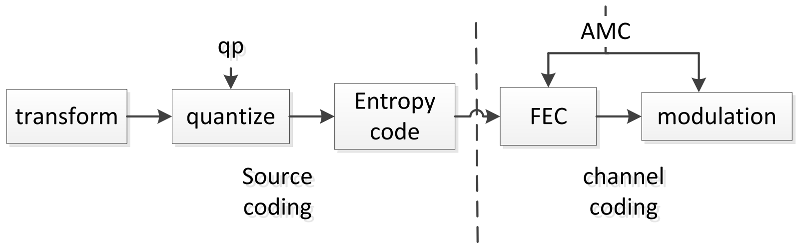

Fig. 1. Traditional transmission system

Video Coding: As shown in Fig.1 typical wireless video uses compress coding technology, including prediction, transform, quantization and entropy coding. In March 2003, ITU-T released H.264 video coding standard. It not only improved the video compression efficiency obviously than ever but also had good network compatibility, especially for those error-prone, easy-blocking IP Internet and wireless mobile network. H.264 compression sits in the application layer. Different quantization parameters produce bit streams with different rates. In order to ensure the video bit rate is not larger than the physical throughput, the sender introduces source rate control mechanism, determining the quantization parameters according to the physical throughput. Notice that the throughput of the physical layer is time-varying, usually in the scale of milliseconds, while video compression time ranges usually in seconds. The mismatch leads to a similar problem of mismatch in the physical layer.

Rateless Coding: A seamless rate adaptive technology is required in the wireless communication, where the transmitter does not need to change the way of transmission and the receiver can automatically adjust as the change of channel condition continuously and smoothly. Assume that the channel noise obeys Gaussian distribution. According to the information theory regarding channel capacity definition, the channel input should obey the Gaussian distribution when the system approaches channel capacity. However the study of the channel coding usually assumes that the channel input signals are evenly-distributed. In order to obtain larger channel capacity, some researchers realize that changing the distribution of the input signal can increase the channel capacity. Forney[7] pointed out that under the Gaussian channel with limited bandwidth, it is impossible to achieve the channel capacity if modulation uses equiprobable constellation diagram and there will be about $\pi e/6$ dB performance loss. Overlay mapping is proposed as a kind of implicit shaping technique initially. Duan[8] proposed to superpose output of several independent code words together with this principle in the mind. According to the central limit theorem, superimposition code at the receiving end approximately obeys Gaussian distribution. Afterward, Ma and Ping[9] further studied this constellation mapping method and named it sigma-mapping. Recently, sigma-mapping was revisited by Wo[10] and Hoeherp[11], and was renamed as superposition mapping(SM). They also investigated posterior probability detection based on SM and LDPC encoding strategy. RCM[3] is essentially an iterative sigma mapping. It sums up the product of L bits. Every L bits encodes into one modulated symbol with the corresponding weights:

$$y_i = \sum_{l=1}^{L} w_l \cdot x_{i_l} \tag{1}$$

where x represents the bit sequence, L is the load factor, W = w_1, w_2, ... w_L, $w_L \in$R, are weight values. Each weight value should be present once in one arithmetic sum. Subscript i_l represents corresponding bits index of w_l in the modulation symbol set y_i. It is important to note that the sigma mapping mentioned in the literature [9] and [10] must first transform coding bits into the BPSK modulated symbols before calculating weights sum. While RCM calculates binary bit arithmetic weights sum directly. Because unipolar signal efficiency is not high, RCM uses the positive and negative symmetric weight in order to produce positive and negative symmetric symbols, avoiding the occurrence of unipolar signals.

III. The Framework

Fig. 2 provides an overview of the proposed framework. Like H.264, COQRC makes use of prediction, transformation and entropy encoding. The difference between predicted value PRED and true value is transformed and quantized into a group of transformation coefficients. These coefficients generates compressed bit streams, along with some other information required for decoding, such as prediction parameters, the motion vector, etc. The quantization parameters are determined by bandwidth and spectrum efficiency. Section IV gives a detailed description of generation method.

The transmitter produces modulated symbols according to RCM strategy. It sends a small amount of the symbols to the receiver at first. If the receiver decodes the message correctly it stops sending symbols. Otherwise, the transmitter increases the symbols by degrees until the receiver decodes the bitstream correctly and sends feedback to the receiver. Assume that a single discrete source $\mathbf{u} \in GF(2)$ transmits over the AWGN channel whose capacity $C = B * \log_2(1 + S/N)$. When the transmitter receives feedback ACK, it will stop sending symbols. During the whole process, the transmitter does not need to know the channel condition.

The client receives symbols successively and uses the received symbols to decode bits. After the user decodes bits correctly, it sends feedback to the sender. The client decodes

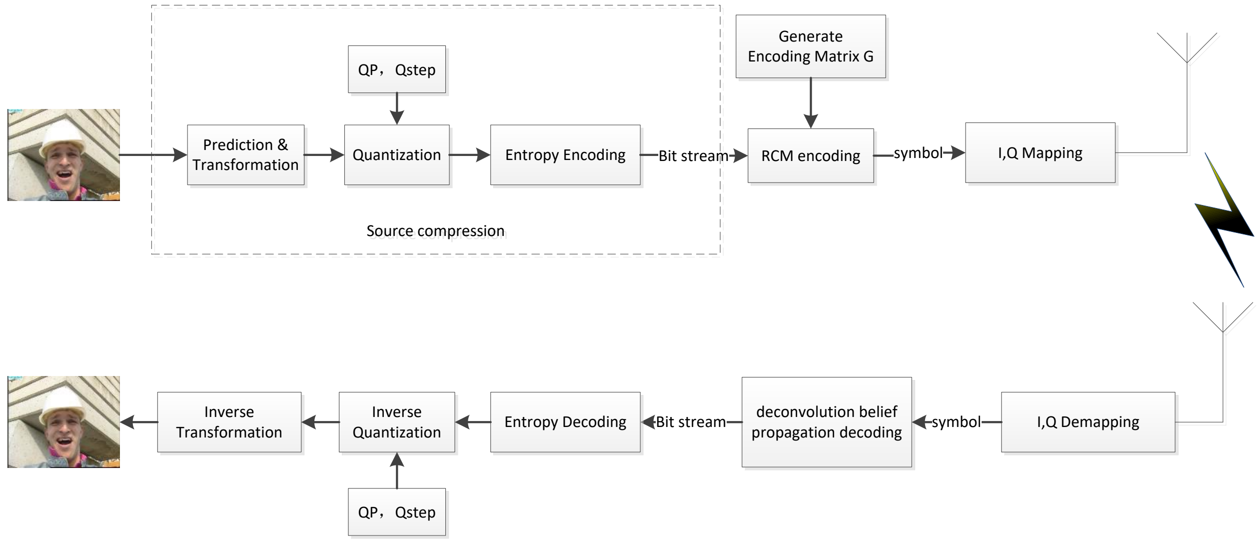

Fig. 2. Block diagram of COQRC

received symbols with de-convolution belief propagation algorithm. Section V illustrates the realization of de-convolution belief propagation algorithm.

After the receiver decodes symbols into bit streams, it performs the reverse operation of the transmitter. The bit streams are entropy decoded into quantized coefficients first. Then, it performs inverse quantization and inverse transformation.

IV. Video Sender

It turns out that in COQRC the video quality performance is a function of both the quantization parameter and the channel capacity $C = B * \log_2(1 + S/N)$. In practice, the quantization parameters Q and values of C are computed ahead, and each pair of optimized values (Q, C) is stored in a Lookup Table (LUP).When a video is to be transmitted, the encoder chooses the appropriate (Q, C) pair closest to the estimated channel capacity. This approach, based on different rate codes, is conceptually like Adaptive Coding and Modulation (ACM), currently used in wireless communication systems such as LTE or 802.11n[12]. Though ACM is used to adapt the coding rate to the channel, in the COQRC it is mainly used to adapt to the quantization parameters.

The quantization parameter (QP) is determined by the spectrum effectiveness (SE) of channel coding as well as the channel bandwidth. QP influences compression ratio and quality of the image. The principle of the scalar quantization is as follow:

$$FQ = round(\frac{y}{QP}) \tag{2}$$

where y is the input samples, QP is quantization parameter, and FQ is quantized value. The inverse quantization is as follow:

$$y' = FQ \cdot QP \tag{3}$$

During the process of quantization and inverse quantization, QP plays an important role in the compress ratio and image precision. If QP is large, the range of quantized value FQ will be small. Accordingly, the coding length after compress will be short. However, it loses more information after inverse quantization. On the contrary, if QP is small, the FQ range will be large and the coding length will be long. It retains more information after inverse quantization. The encoder should adjust its QP according to image content and channel condition automatically. The encoder needs to balance between coding length and image precision in order to achieve the overall good quality.

The length of bit streams wireless video transmitting system can be computed using the following formula:

$$N = \frac{SE \cdot M}{2} \tag{4}$$

where M represents the number of symbols which can transmit, SE is the spectrum efficiency. N is the number of bits wireless system can transmit. Further, we will determine QP.

A. Rateless Video Code

RCM is the ideal choice for this application since it produces symbols at all rates. Let N be the size of a code block, K the number of symbols the user needs to decode the code block correctly. When we start to transmit, we transmit K_0 modulated symbols in the beginning. Then increase transmitting symbols with step size K_c until the user decodes the message correctly. Hence the transmission rate is defined as:

$$R = N/K = N/(K_0 + \mathrm{i} \cdot K_c) \tag{5}$$

It is required to distribute bit energy to symbols uniformly in order to achieve rateless. The rateless coding and modulation requires that each bit is duplicating sampled by more than one symbols. Each bits sampling weight has the same Eulerian norm. Only in this way the bit energy increases evenly as symbols gather. We introduce an MN-size encoding matrix G to map from bits to symbols.

1) Encoding matrix construction: In RCM, mapping matrix is low-density. There are L nonzero elements each row. Here, L equals 8. The L nonzero elements are a random order of weight set $\{-1, 1, -2, 2, -4, 4, -4, 4\}$. The self-decodability and channel characteristic should be taken into consideration when construct matrix G. According to Cui[3], encoding matrix row must be regular and matrix column should be regular as far as possible. The weight set should produce modulated symbols as many as possible. Then we construct encoding matrix G. Construct three basic N/8*N/4 matrixes A_1, A_2, A_4. The structure of A_1 is as follow. A_2 and A_4 have the same structure but different nonzero elements.

$$A_1 = \begin{bmatrix} +1 & -1 & & & & \\ & & +1 & -1 & & \\ & & & & +1 & -1 \end{bmatrix} \tag{6}$$

Second, fill encoding matrix G_0 with these three basic matrixes randomly as follow:

$$G_0 = \begin{bmatrix} \pi(A_4) & \pi(A_4) & \pi(A_2) & \pi(A_1) \\ \pi(A_2) & \pi(A_1) & \pi(A_4) & \pi(A_4) \\ \pi(A_4) & \pi(A_4) & \pi(A_1) & \pi(A_2) \\ \pi(A_1) & \pi(A_2) & \pi(A_4) & \pi(A_4) \end{bmatrix} \tag{7}$$

where $\pi(\cdot)$ represents the random arrangement of matrix columns.

Because x equals either 0 or 1, there are 23 different RPC symbols. The symbol values range from -11 to +11. Each two RPC symbols form I (in-phase) and Q (quadrature-phase) components in wireless symbols. In the traditional wireless system, several neighboring bits maps one constellation point. However RCM uses an unconventional 23x23 constellation, denser than the traditional system. When the channel condition is bad, the receiver needs more symbols to decode the bitstream. One bit block in the sender can produce infinite modulation symbols. When the channel condition is good, fewer symbols are enough to decode the message. Otherwise, the sender needs to continuously transmit symbols until it receives ACK from the decoder. The traffic rate is computed by N/K, in which way RCM realized seamless rate adaptation.

V. Video Receiver

The received symbols $\hat{\mathbf{s}} = G * \mathbf{b} + \mathbf{e}$. e represents the white Gaussian noise $e(m) \sim N(\sigma^2)$. The receiver aims to find the optimal solution of the following problem:

$$\hat{\mathbf{b}} = \arg \max_{\mathbf{b} \in \{0,1\}^N} P(\mathbf{b}|\hat{\mathbf{s}}) \tag{8}$$

A. Deconvolution Belief Propagation Algorithm

According to Baron[13], we can solve this problem using belief propagation algorithm. The deconvolution belief propagation algorithm works as follows. Let v and c represent variable nodes and check nodes respectively.

(1) Initialize: use the statistical information as the initial information.

$$u_{v->c} = p_v(1) = p \tag{9}$$

(2) Horizontal Iteration(decode bits with check node): Compute each check nodes probability distribution $p_c(.)$by convolution, shown in Eq. 10. The probability density function of each check node calculates from all of its neighbor nodes. Compute $p_{c\backslash v}(.)$by Eq. 11. Calculate probability density function for each check node c using all of its neighbor nodes except variable nodes v.

$$p_c = (*)_{v \in n(c)}(w(c,v) \cdot p_v) \tag{10}$$

$$p_{c\backslash v} = p_c \tilde{*}(w(c,v) \cdot p_v) \tag{11}$$

Compute $p_v(0)$ and $p_v(1)$ using the probability distribution of noise p_e and the received symbols s_c.

$$p_v(0) = \sum_i p_{c\backslash v}(i) \cdot p_e(s_c - i) \tag{12}$$

$$p_v(1) = \sum_i p_{c\backslash v}(i) \cdot p_e(s_c - i - w(c,v)) \tag{13}$$

At last, calculate and normalize the message $u_{c->v}$:

$$u_{c->v} = \frac{p_v(1)}{p_v(0) + p_v(1)} \tag{14}$$

(3) Vertical Iteration(decode bits with variable nodes): for each variable node, calculate $p_v(0)$ and $p_v(1)$ by multiplication.

$$p_v(0) = (1-p) \prod_{u \in n(v)} (1 - u_{u->v}) \tag{15}$$

$$p_v(1) = p \prod_{u \in n(v)} u_{u->v} \tag{16}$$

Calculate and normalize the message with message from each neighbor check node. Each iteration repeats step 2 and step 3.

(4) Hard decision: Use above equations to decide the variable node.

Fig.3(a) is spectrum efficiency comparison between RCM and Turbo used in LTE. The number of iterations used in deconvolution belief propagation influences the accuracy of decoding as well as the complexity of decoding.

VI. Implementation and Evaluation

A. Experimental Setup

In COQRC, the main factors influencing the video quality are SNR, quantization parameters, the spectrum efficiency of channel coding and bandwidth. Eq. 4 indicates the relation between the spectrum efficiency and bandwidth. In this section we compare video quality of COQRC and mobile video transmitting over LTE, which uses Turbo as its channel code. Noticed that when SNR is low, the spectrum efficiency of RCM is inferior to Turbo, we replaced RCM with A-BICM near those SNR.[14]

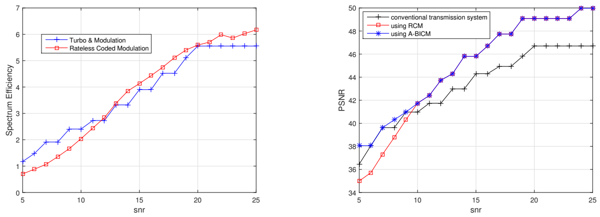

Fig. 3. Performance evaluation: (a)Spectrum efficiency comparison. (b)PSNR comparison.

Channel setting: The channel used is AWGN with SNR ranging from 5dB to 25dB. The error rate is proportional to the received RCM symbols number. The more RCM symbols received, the lower the error rate is. Without bandwidth constraints, every bit can be correctly decoded. To fairly compare, we have limited bandwidth the same as Softcast [15]. For example, the sender sends 50688 symbols for a 352*288-size video sequence in softcast. In that way, COQRC using RCM can only send 50688 symbols for a same-size video. Using the number of symbols along with spectrum efficiency we figure out the quantization parameters. The bit stream after entropy encoding is sent within the bandwidth.

Reference schemes: We use LTE as the reference scheme. The channel coding used in LTE is 1/3 rate Turbo Coding. The modulation schemes are QPSK, 16QAM and 64QAM.

Evaluation criteria: Peak Signal-to-Noise Ratio (PSNR) is usually used to measure video/image quality. We compare these two schemes in PSNR, which is computed with the following equation:

$$PSNR = 10\log_{10}\frac{255^2}{MSE} \tag{17}$$

where MSE is the mean square error of all pixels. For the video, it takes the average value of all frames. The key thing to note is that a $PSNR$ below 16dB is effectively noise, whereas a $PSNR$ above 40dB is excellent quality.

B. Comparison With The Reference Schemes

In the decoder, the iterations times influence accuracy and complexity. In our system the decoder iterates is set to 30.

We choose four standard test video sequences of different resolutions. Fig.3(b) shows the PSNR comparison between COQRC and traditional digital system when the video sequence is park_run.yuv of 720p.

VII. Conclusion

In this paper, we analyzed the spectrum efficiency and the reliability of rateless coded modulation scheme in wireless video transmission system and designed a transmission system based on RCM. It was confirmed that the proposed scheme saves bandwidth which comes from the flexibility of RCM. The optimal quantization parameter setting problem is formulated in the paper. Experiments have validated the feasibility of RCM in practical systems. Performance comparison shows that the proposed system outperforms conventional video transmission system by 2-3dB in PSNR.

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