Modeling of Adaptive Wireless Link for MPEG-4 Video Transport in UMTS Network

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***Abstract* — Conventional transmission schemes perform poorly in wireless channels. In this paper, an analytical adaptive model for wireless links via heuristic TCP/FEC functions is proposed with chosen modulation format for transporting mobile MPEG-4 video. A wireless channel introduces that the bit errors at the link-layer packets are due to the time-invariant features over UMTS network and consequently the optimal perceived video quality at the client can be improved when adaptive TCP throughput after error-corrections is achieved for the certain predefined system threshold of the residual packet error rate. The simulation results show that a video quality can dynamically adapt to the optimal FEC codes at an appropriate selection for modulation scaling in the radio-link layer.**

I. INTRODUCTION

The demand for high data rates and quality of service (QoS) is growing at a rapid pace with the advent of 3G/3.5G cellular networks technologies of the Universal Mobile Telecommunications Systems (UMTS). It enables multimedia applications (e.g., medical video streaming in Telemedicine systems, video conference, Video on Demand) access via the radio interface [1-4]. Specifically, Wideband Code Division Multiple Access (WCDMA) allows mobile Internet users to access a variety of multimedia contents available on the Internet in a seamless fashion, i.e. always on, at data rate up to 384 kbps in wide-area coverage and high user mobility. In addition, CDMA offers more flexibility than Time-Division Multiple Access (TDMA) to cross-layer design for resource-allocation due to its capability to support simultaneous transmissions [1][3]. However, the performance of wireless links is often prone to errors due to various physical impairments of the highly noise or deeply fading. Therefore, the key problem in this paper is to provide a robust cross-layer model for mobile MPEG-4 video quality performance over UMTS network.

On the other hand, wireless multimedia networks often cannot provide the guaranteed quality of service (QoS) metrics of data loss, delay, and throughput. Many challenges of video traffic are encountered on the wired and wireless Internet links [5-8]. Some challenges deal with high packet loss rate due to the congestion of buffer overflow over wired networks; and others are mainly caused by the characteristic of wireless links. More precisely, there is not only scarcer and expensive wireless resource (bandwidth and power) than their wired counterparts, but also there exists a high bit error rate in wireless channel which is caused by multi-path fading, intersymbol interference and noise disturbance and eventually resulting significant effect on multimedia transmission [9-10]. In particular, the link errors may result in packet (segment) losses, and TCP sender interprets such loss as a signal of network congestion and consequently decreases the transmission rate [10-16]. To promote robust and high video quality, there exist several recent adaptive transmission schemes either to optimize rate control using for example a form of real time reinforcement learning known as Q-Learning as in [2], or to improve the link reliability observed by TCP equation model as in [10,11]. These solutions may involve jointly modeling of the link adaptation to support the QoS metrics through a cross-layer design using various approaches such as adaptive rate control [11], adaptive selective Repeat (ASR) protocols (re- transmission) [12], adaptive modulation coding (AMC) [13], and adaptive forward- error-correction (FEC) [14-16].

In this paper, we propose an analytical model to improve the wireless link reliability via the heuristic TCP/FEC functions with chosen modulation format for transporting mobile MPEG-4 video. The analysis introduces adaptive link model for the bit errors of the time-invariant wireless channel over UMTS network; and then at the predefined threshold of the residual packet error rate the perceived video quality of the client can be optimized using adaptive TCP throughput after error-corrections. The optimal allocation bandwidth required for a high video transport is evaluated by achieving the optimal code efficiency of byte-level RS FEC and/or CRC SR-ARQ scheme when an appropriate modulation scaling and predefined system threshold of the residual packet error rate are carefully selected under different channel state estimations. The simulation results reveal clearly that a video quality can be dynamically achieved by verifying optimal FEC codes at a certain modulation scaling in the radio-link layer.

II. SYSTEM MODEL

We consider the network architecture of the UMTS which is designed for only packet-switched operation based-IP at the end-nodes, namely, the mobile client and server. The MPEG-4 video is streamed [17] whereas the video input needs to be encoded by codec (media server) to form bitstream and is then transmitted to the network. At the decoder side, the video is received first by the decoder and then displayed on the terminal. A block diagram of such video transmission system is shown in Fig. 1 which consists of a transmitter (base station), a receiver (mobile user) and a communication channel with a limited bandwidth. In this system, we consider the wireless

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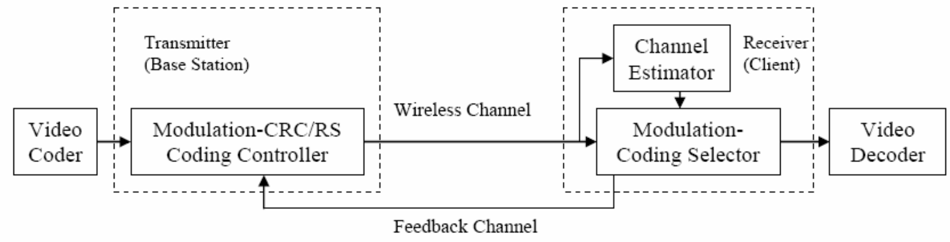

Fig. 1 Adaptive wireless link configuration

channel capacity is limited by the bandwidth B_w. In addition, the effective residual packet loss rate ε_s is only arising due to the corruption of bit errors ignoring the congestion of opening any concurrent TCP video connections on the same channel or the congestion of wired links. Specifically, the residual error probability of wireless link is still essential parameter for a perceptual video quality at the client end (mobile client). Therefore, we focus on the Acknowledged Mode of Radio Link Control layer (RLC/AM) because this protocol performs error detection and recovery for applications requiring a reliable channel [1] [7] by retransmitting erroneous RLC blocks (packets) at the expense of transfer delay. The retransmission strategy adopted by the AM is the Selective-Repeat ARQ (Automatic Repeat reQuest) scheme where the RLC blocks retransmitted are those that receive a negative acknowledgement.

To provide more robust system, channel state information (CSI) can be estimated at the receiver and fed back to the transmitter controller for selecting adaptive modulation and/or channel coding such as CRC/RS codes [7]. However, a packet of RLC consists of a header and a payload which carries higher layer data. An erroneous packet is discarded by RLC, which would result in an incomplete IP packet and consequently it can not be delivered correctly to higher layers. The transmitter constructs small packets (blocks) of K bits, and transmits in a continuous stream. To ensure that bits received in error are detected the transmitter attaches a C bit CRC to each data packet, making the total packet length $K + C = L$ bits as in Fig. 2. Moreover, each RLC packet length L is also encoded using *inner code* protection RS FEC coding. Hence, encoded packets by concatenated CRC/RS FEC codes are transmitted through the air and processed by the receiver. The RS FEC decoders at the receiver are assumed to be able to detect all the errors in the received packets. Upon decoding the packet, the receiver sends an acknowledged, either positive (ACK) or negative (ACK), back to transmitter via ARQ scheme [12]. We assume this feedback packet goes through a separate control channel, and arrives at the transmitter instantaneously and without error. If the CRC decoder detects any error and issues a NACK, the transmitter uses selective repeat protocol to resend the packet. It repeats the process until the packet is successfully delivered. On the other hand, having employed an adaptive byte-level RS (N, K) FEC code at the data-link layer, an error-resilient video system can be established through the dynamic improvement in the link reliability in terms of the residual packet loss rate after correction at the client.

Header (16 bit)	Payload $\geq K$ (bit)	Trailer CRC-16

Fig. 2 The link layer format

III. Proposed Approach

A. TCP over Wireless

Although the complex behavior of TCP is due to its various mechanisms such as slow start, congestion, timeout, etc., it has been shown in [11,18] that the throughput of a TCP-Friendly connection is a simple expression in the absence of timeouts. Hence, the steady state TCP goodput (i.e. maximum throughput) of a long-lived connection can simply be obtained by scaling the throughput by a factor of $(1-\varepsilon_s)$

$$\mathrm{T}_{TCP,s} \approx \min\left(W_{\max}\,,\ \frac{k\cdot S}{\sqrt{\nu\varepsilon_s}} \right) \frac{(1-\varepsilon_s)}{RTT} \ \leq\ B_w \qquad (1)$$

where $W_{\max}$ is the maximum congestion window size of the TCP sender, S is the packet size (MSS), ν is the number of packets (segments) that are acknowledged by a received ACK, ε_s is the residual end-to-end packet (segment) loss rate for wireless client, k is a constant that is usually set to either 1.22 or 1.31, depending on whether the receiver uses delayed acknowledgments, and RTT is the round trip time experienced by the connection per packet sent [14]. For sake of simplicity, we use the fixed-point end-to-end analytical TCP model shown in Fig. 3 for adaptive wireless link [16]. The output of this model in our proposed approach emphasizes on the key parameter ε_s with assumption of fixed round-trip time (RTT). Then the resultant traffic in terms of arrival rate can be evaluated as the effective TCP throughput in this model.

B. SR-ARQ Throughput

For conventional pure or hybrid Type-I SR-ARQ with fixed modulation, the throughput efficiency can be defined as [19,20]

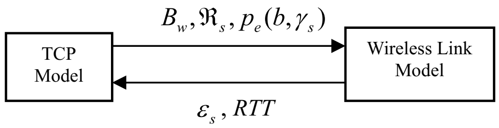

Fig. 3 The fixed-point analytical model

$$\eta_{SR} = P_C \cdot \left(\frac{K}{K+h} \right) \tag{2}$$

where P_C is the probability that a received packet with $N = K + h$ contains no error, and $h = \beta + \tau$ denotes the number of bits at header and trailer in the packet. Thus throughput η_{SR} is a function of the modulation mode, packet size and received SNR. In particular, we investigate the optimal packet size based modulation mode in order to address SR-ARQ and a hybrid SR-ARQ FEC which both are capable of increasing the effective throughput of the system [21-22].

Let $\Re_s$ with $\Re_s \le B_w$, be an available symbol rate (bandwidth) at radio-link layer for a given TCP connection. Each TCP packet (segment) *MSS* is divided into small packets (blocks) of length *L* bits, where $L = K + \beta$, and β denotes the header bits. Each block is transmitted symbol by symbol through the channel and corresponds to $L/b = L_s$ MQAM symbols. Each MQAM symbol is modulated using fixed power MQAM. Hence, the maximum channel throughput in (bps) can be roughly represented at the hardware-link layer for any modulation scheme as follows [23-24],

$$B_{Phy,m} = \Re_C \cdot b\Re_s \cdot f(b, \gamma_s, L) \tag{3}$$

We will refer to the term "*mod m*" to indicate to a specific choice of an adaptive modulation mode, and $\Re_C$ is a redundancy ratio, $\Re_C = (L-h)/L$. L is a packet length (in bits) where each block is transmitted bit by bit through the channel. $f(b, \gamma_s, L)$ denotes an appropriate function for the probability P_C in (2), which may describe link errors by providing the impact of the channel signal-to-noise ratio per symbol and the packet length on TCP throughput performance of (1). However, other definition can be found in [10] in terms of round-trip time and link errors. Then, $f(.)$ can be considered as the packet success rate (PSR) to define the probability of receiving a packet correctly, and γ_s is the SNR per symbol given by

$$\gamma_s = E_s / N_o = P/(N_o \Re_s) \tag{4}$$

For any modulation scheme, E_s, N_o, and P represent the symbol energy, the one-sided noise power spectral density, and the received power respectively. Furthermore, $f(.)$ in (3) can be expressed as

$$P_{Loss}^{CRC}(b, \gamma_s, L) = 1 - f(b, \gamma_s, L) = 1 - [1 - p_e(b, \gamma_s)]^{L/b} \tag{5}$$

Based on the symbol error rate $p_e(b, \gamma_s)$, the PER for ARQ with only CRC-16 error detection mechanism can be calculated. Assuming all possible errors in a packet can be detected, the PER of a packet with a payload *L* bits is given by (5). P_{Loss}^{CRC} is a packet loss rate due to link bit errors. Thus we can define $P_{Loss}^{CRC}(b, \gamma_s, L)$ as the probability of packet (block) error in terms of packet length in L bits, symbol length *b* and γ_s which is being SNR per symbol for modulation scheme.

To increase the link reliability, there exist two approaches: retransmission (ARQ) and forward error correction (FEC). In this section, we consider a CRC- Selective-Repeat SR-ARQ scheme. Assuming that the CRC can detect every possible packet errors and neglecting the messaging overhead, the throughput of (3) using SR-ARQ can be bounded by [20][24]

$$B_{SR-ARQ,m} \le \Re_C \cdot b\Re_s \cdot \left(1 - P_{Loss}^{CRC}(b, \gamma_b)\right)^{L/b} \tag{6}$$

On the other hand, when the maximum packet size is controlled relative to the prediction BER in wireless channel the network throughput may significantly improve [25-26]. By assuming L has continuous values, then differentiating (3) with respect to L and setting it to zero yields L^* [24],

$$L^* = \frac{h}{2} + \frac{1}{2}\sqrt{h^2 - \frac{4bh}{\ln(1 - p_e(b, \gamma_s))}}\ , \tag{7}$$

From (7), we can estimate the optimal payload size *without* FEC coding at certain time of design to increase the throughput of SR-ARQ in (6). The optimal packet length L^* depends on the constellation size 2^b, the SNR per symbol γ_s, and the probability of symbol error p_e of MQAM which is defined (approximately) in AWGN channels by [27],

$$p_e(b, \gamma_s) \le \frac{4}{b}\left(1 - 2^{-b/2}\right) Q\left(\sqrt{\frac{3}{2^b - 1}\gamma_s}\right), \tag{8}$$

$Q(.)$ is Gaussian cumulative distribution function.

In non-fading or slowly fading channels where the fade duration is longer than the packet period, the system throughput and its optimization can also be achieved. In this case, the packet error in burst-error condition cannot easily be modeled by a single equation. The reason is that the distribution of error bits is not uniform. To simplify the estimation of BER performance, a BPSK scheme (b=1) over AWGN channel can be applied for upload/download streams and the probability of bit error can be given as function of SNR per bit (γ_b),

$$p_e(\gamma_b) = Q\left(\sqrt{2\ \gamma_b}\right). \tag{9}$$

C. Optimal FEC Code

We consider an (N,K) Reed-Solomon code to represent the level of redundancy at byte information. An $(N\text{-}K)$ parity symbols are added to N data symbols to form a codeword of size N. The number of code length N is *fixed* and the information symbols per codeword K is *varied* to adjust the redundancy level of the code. Here a symbol is the basic information unit used in a RS code, and is composed of certain number of bytes (or bits).

Now, to generate the byte-level FEC, the encoder processes in symbols, where each symbol consists of m bits (m=8 in general). Assume a block size is being N_b (in byte) and K_b (in byte) with $K_b \geq 1$ of source data are packed with N_b - K_b parity bytes, where $K_b = N_b$, N_b-2…., then RS(N_b, K_b) code is able to correct up to byte errors in a packet,

$$t_b = \lceil (N_b - K_b)/2 \rceil \tag{10}$$

The block (small packet) size N_b is limited by 2^m-1 bytes; therefore, for m=8, $N_b \leq 255$. To provide adaptive FEC scheme, we consider N_b represents the maximum block length ($N_b = L_{max}$) in byte, whereas extending the total block length can be valid to $(K+B) \leq L_{max}$, where B is attached error corrections in bytes expressed as,

$$B = 2t_b = L_{\max} - K_b \tag{11}$$

Since the RS(N_b,K_b) code corrects up to t_b errors, the probability that a random packet cannot recovered by byte-level FEC is given by [6],

$$\varepsilon_s(b,\gamma_s,N_b,K_b,t_b) = \sum_{i=t_b+1}^{N_b} \binom{N_b}{i} \left[P_B(b,\gamma_s)\right]^i \cdot \left[1 - P_B(b,\gamma_s)\right]^{N_b - i} \tag{12}$$

where $P_B = 1 - \left[1 - p_e(b,\gamma_s)\right]^m$ and m=8.

Since a packet can be larger than the block length N_b especially where small block lengths are used, the packet loss rate (PLR) for FEC is given by

$$PLR^{FEC}(S,b,\gamma_s,N_b,K_b,t_b) = 1 - (1-\varepsilon_s)^{\lceil S/K_b \rceil} \tag{13}$$

where $\lceil . \rceil$ is the ceiling function of the number of blocks required to send S bits. In our model, we assume that a TCP packet size is equal to the block size and consequently the residual packet loss rate can be deduced as $PLR^{FEC} = \varepsilon_s$.

Given the feedback from the end client, the sender has to decide the byte-level FEC rate (number of byte parity bits per second, i.e., the *inner* rate $\Re_{inner} = K_b / N_b$) that needs to be adapted with its original transmission rate $\Re_s$, including all the redundant bits. Thus we study the optimal throughput as follows:

Let $\Gamma_{C,w}$ be the maximum throughput (goodput) of the wireless client *after* only error detection and without error correction,

$$\Gamma^*_{CRC,w} = \eta^*_{CRC} \times b\Re_s \quad \leq B_w \tag{14}$$

where $\eta^*_{CRC} = \Re^*_C \times r^*_{link,CRC}$ is being the optimal code efficiency when only CRC error detection is employed [25]. The redundancy ratio is expressed as, $\Re^*_C = (L^* - C - 16)/L^*$ and $r^*_{link,CRC} = \left(1 - p_e(b,\gamma_s)\right)^{L^*/b}$ represents the optimal link reliability based only SR-ARQ scheme.

In other words, when no FEC coding is applied, the CRC-16 is used to provide the optimal packet length which consequently introduces the maximum throughput over network. However, a typical Automatic Repeat Request (ARQ) packet is adopted the header β of 16 bits. This could be a big overhead in short packets (e.g., 640 bits for WCDMA). Since the delay is proportional with the packet length, so the optimal packet length needs to be fit with the required packet-length in WCDMA system [4].

To study FEC allocation problem at byte-level, we consider the following procedure: For a given N_b we can find the optimal K_b in order to maximize the end-to-end goodput of the mobile client,

$$\Gamma^*_{C,w} = \eta^*_{coding} \times b\Re_s \quad \leq \quad B_w \tag{15}$$

and the optimal RS FEC code efficiency can be deduced as,

$$\eta^*_{coding} = \Re^*_{inner} \times r^*_{link} = \left(L_{\max} - h\right) \times r^*_{link} / L_{\max} \tag{16}$$

where h=C+16+9$B/2$ represents the number of overhead bits per packet at data-link layer [24,28], and $r^*_{link} = \left(1 - \varepsilon^*_s(b,\gamma_s,k^*_b)\right)$ denotes the optimal link reliability s*uch that* the end-to-end packet loss rate after error correction is no more than certain predefined threshold value, ε_o (say, 1%-3%) for loss rate over all clients. Further, $b\Re_s \leq B_w$ and $\Gamma_{C,w} \leq B_w$, where B_w is being a limited wireless channel capacity.

In our scenario, we consider there is no packet loss due to congestion (overflow buffer), then (12)-(16) can only be used when packet errors are due to the wireless channel feature. To reduce the complexity of optimization problem, we use a simple two-step algorithm [6] for a given certain residual system threshold of packet loss rate ε_o, the maximum coded block length L_{max}, b, γ_s, and C.

Step (1): The channel status is first estimated by CSI estimator at the receiver in terms of the bit error rate $p_e(b,\gamma_s)$ via channel feedback response.

Step (2): The residual packet loss rate is accounted for only ARQ scheme using CRC detection, then if the residual loss is less than ε_o, no adaptive FEC is required. Otherwise Byte-level FEC optimization is followed to find K_b^* such that the largest $K_b \le N_b$ and then ε_s in (12) should be no greater than ε_o. As a result, optimal t_b can be achieved to provide a required FEC code (N_b, K_b^*).

D. Adaptive TCP/ FEC

Using the fixed point analytical approach based adaptive FEC in Fig. 3, we can adapt the required optimal allocated bandwidth as follows: Given the estimate of ε_s in (12) and a certain value of *RTT*, we can compute the available TCP goodput ($T_{TCP,s}$) using (1). Then, to achieve the optimal TCP-adaptive FEC for wireless clients, the real TCP goodput ($\mathrm{T}_{TCP,eff}$), i.e., the optimal allocated bandwidth required for TCP protocol, can be accounted for the minimum of the achievable TCP-Friendly goodput ($\mathrm{T}_{TCP,s}$) and the effective link bandwidth ($\Gamma_{C,w}$), as in [16],

$$\mathrm{T}^*_{TCP,eff}(K_b^*, b, \varepsilon_s) = \min\left(\Gamma_{C,w}(K_b^*, b, \varepsilon_g), \mathrm{T}_{TCP,s}(K_b^*, b, \varepsilon_s)\right) \quad (17)$$

Equation (17) is being subjected by a network bandwidth constraint that $B_{Phy,m} \le B_w$ and $\mathrm{T}_{TCP} \le b\Re_s < B_w$. As a result, the optimal FEC code (N_b ,K_b^*) is the code that maximizes TCP throughput after error correction at the client (i.e., maximum TCP throughput), and is computed as

$$K_b^*, \varepsilon_s^*, b = \arg\max_{K_b \in N_b, \varepsilon_s \in p(b,\gamma_s)} \mathrm{T}_{TCP,eff}(K_b, b, \varepsilon_s) \quad (18)$$

In fact, the TCP throughput is maximized when the achievable TCP throughput equals the effective channel bandwidth. Ideally K_b is just the solution to the equation $\Gamma_{C,w} = T_{TCP,s}$. Thus the protocol will be based on the client feedback to the base station and use a lookup table at the base station which is a *priori* generated to find the FEC code that yields the largest goodput as a function of channel SNR estimate. However, in case of no adaptive FEC is required, we can evaluate the effective TCP throughput using (17), but with an adaptive optimal packet size of SR-ARQ scheme defined in (8).

IV. Video Quality Estimation

To evaluate the video quality performance, the following scenario is applied:

- A flow is considered with data rate (throughput) not exceeding the maximum data rate of TCP connection in the same network conditions [11].
- There is no extra-traffic or congestion due to concurrent TCP video connections. The packet loss is only due to channel bit errors, and the minimum round-trip time is being at certain value. The only restriction is wireless channel capacity which should imply $\mathrm{T}^*_{TCP,eff} \le B_w$. The backward route from the client to video server is assumed to be congestion-free but not error-free due to bit errors.
- Optimal control rate should result in the highest (optimal) possible throughput and the lowest residual packet loss rate after adapting a required optimal packet size of SR-ARQ scheme in (7) or optimal FEC code in (18) *w.r.t.* the predefined threshold value ε_o at fixed suitable modulation through b. To avoid any network instability, $b\Re_s$ is regarded as the available bandwidth for video source in bit per second and by adjusting the video traffic, the high-quality video play-out at the receiver can be predicted. Hence, for under-utilized channel, $\mathrm{T}^*_{TCP,eff} \le b\Re_s < B_w$ holds when only one TCP connection exists for a single user.
- Using adaptive TCP/FEC heuristic scenario, the predicted optimal playable frame rate (PFR) based on maximum effective TCP throughput in (17) can be evaluated as in [8],

$$PFR^* = G.W_I \cdot [1 + \chi_P + N_{BP} \cdot W_B(\chi_P + W_I \cdot W_P^{N_P})] \quad (19)$$

$$\chi_P = \left(W_P - W_P^{N_P+1}\right) \times \left(1 - W_P\right)^{-1}, \ W_i = \left(1 - \varepsilon_s^*\right)^{S_i}, \quad (20)$$

$$G = \mathrm{T}^*_{TCP,eff}.L_{\max}^{-1} / \left(S_I + N_P S_P + N_B S_B\right). \quad (21)$$

W_i stands for the successful transmission probability of the *i-th* frame type (I, P, and B) in a GOP(N_P, N_{BP}) pattern taking into account the end-to-end packet loss rate after error correction given in (12). N_P, the number of P-pictures in a GOP, and N_{BP} the number of a B-pictures between I and P pictures (the distance between I or P pictures). In consequence, the total number of B-pictures can be deduced as $N_B = (1 + N_P) \times N_{BP}$. S_i denotes packet size of the *i-th* frame type. In our analysis, L_{max} (in bytes) must be adapted to CSI estimation using byte-level FEC coding or CRC SR-ARQ scheme, $\mathrm{T}^*_{TCP,eff}$ is defined as the effective network throughput received at the client in (bps), i.e. the optimal allocated bandwidth required for TCP protocol under the constraint of (18). G denotes the optimal number of GOPs per second. S_I, S_P, and S_B are the frames' sizes of the I, P, and B frames in GOP pattern (in packets), respectively.

V. Simulation Results

This section describes the performance of MPEG-4 video streaming over UMTS using our proposed approach. The performance evaluation is carried out by considering the transport Dedicated Channel (DCH) which is reserved for only a single user [3]. The effective throughput (allocated bandwidth) estimation is based on TCP/adaptive FEC and the source video rate, i.e. symbol (bit) rate in AWGN channels. Table I is used to describe the network parameters required for MPEG-4 video stream over WCDMA network in UMTS. We assume that a TCP packet size (MSS) is equal to the maximum length of packet (block) at the data link layer as in [14]. The results were conducted using Matlab programming with three RLC packet lengths 32, 64, and 80 bytes for UMTS [4] when the system threshold of residual packet error (loss) is defined at the video server (Base station).

First the RS FEC scheme is considered at the worst channel state estimation for the BPSK modulation when the system threshold at the base station controller is setting with ε_o=1%. Fig. 4 displays the optimal FEC code efficiency (maximum channel throughput) under various packet length values for error-correction codes. It is found that the efficiency significantly achieves higher values at low bit error rates; and in contrary as the packet length increases to 640 bits (80 bytes) the highest optimal code efficiency can be achievable.

Fig. 5 depicts the optimal packet length vs. channel SNR/bit in AWGN channels when only CRC SR-ARQ scheme is employed under various values of modulation scaling *b*. Note that, as *b* is increased along the scale of channel SNR, the optimal RLC packet length can be obtained with maximum length no more than 640 bits to fit with the typical WCDMA packet length in [4]. As a result, optimal video quality in terms of frames per second can be achievable in Fig. 6 under variable modulation scaling depending on the CSI estimations over the wireless links. This is so-called the wireless link adaptive mode. In contrary, adaptive FEC is applied using various error correction codes. Figures 7 and 8 reveal the optimal video quality of the adaptive link mode (adaptive FEC codes) for various packet length values. The link adaptation is obtained at the worst channel state when BPSK (*b*=1) and for various values of *b* is used. It is clearly shown that the optimal PFR can be dynamically adapted to the CSI estimation compared to the results of Fig. 6; and in contrary the optimal required correction bytes increase as far as the channel bit errors increase. For example, in Fig. 7 the codes of RS(80,78), RS(80,76), RS(32,30), and RS(32,28) provide roughly the full motion play-out frame rate at low channel errors of 1×10^{-4} to 1×10^{-3}; meanwhile, RS(64,62) or (64,60) introduces video scaling of 24 [fps]. On the other hand, the code overhead will considerably increase at high channel errors, e.g., 5×10^{-2}, and in consequence the optimal video quality will not be improved more than 10 [fps] on average.

To investigate the impact of modulation scaling *b* in our proposed cross-layer model, Fig. 8 explains clearly the link adaptation mode for two RLC packet lengths. At low channel SNR/bit values (i.e., less than 12 dB) the *value of b* is no more than 4 in order to achieve the optimal video quality (e.g. 10-30 fps). Otherwise, the good state of CSI estimation will allow the transmitter controller providing higher values of *b* with higher bit rates to support the high video source rates at transmitter (video server) and consequently generating the shorter reception time for the perceived video quality at the client end, where the reception time is roughly equal $T_{rx}=(L_{max}/b.R_s)$ *neglecting* the effect of RTT_{min}.

Table II tabulates an example comparison of *b* effect on the video quality under only CRC SR-ARQ scheme without adaptive FEC coding. Note that the maximum packet length at data link layer and transport layer is the same in all the cases.

TABLE I
SIMULATION PARAMETERS FOR MPEG-4 VIDEO STREAM OVER UMTS NETWORK

Physical Channel	AWGN Channel
B_w	1Mbps (1xRTT CDMA)
Modulation Scaling	*b*=1 (BPSK), *b*=2,4,6,8 (M-QAM)
γ_s	1…25 [dB]
R_s	100 ks/s
Data-link Layer	RLC DCH CRC-16(AM mode), RS FEC
$S_{,}\ L_{max}$	32, 64, 80 byte (for WCDMA)*
ε_o	1%-5%
Transport Layer	k =1.22, RTT_{min}=168 [ms], ν =1,
Application Layer	MPEG-4
F_o (video source)	30 [fps]
GOP(2,3)	*I-BBB-P-BBB-P-BBB*
(S_I, S_P , S_B)	Typical values (25,8,3) [packet]

* We assume that a TCP packet size (MSS) is equal to the maximum length of packet (block) at the data-link layer for *b*=1 [14].

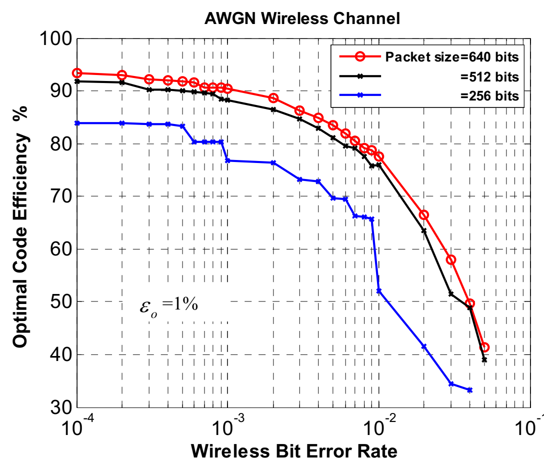

Fig. 4 Optimal code efficiency using RS FEC code under various error conditions and for BPSK scheme over AWGN wireless channel

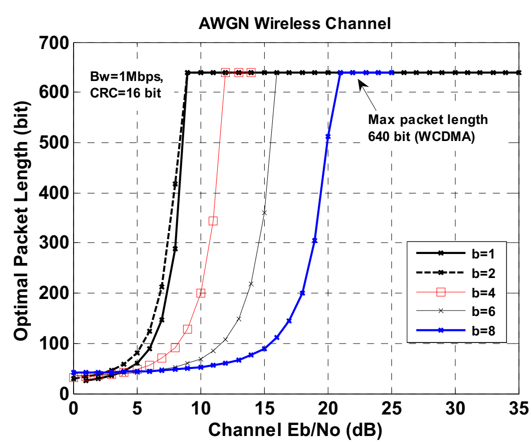

Fig. 5 Optimal packet length vs. channel SNR/bit in AWGN channels

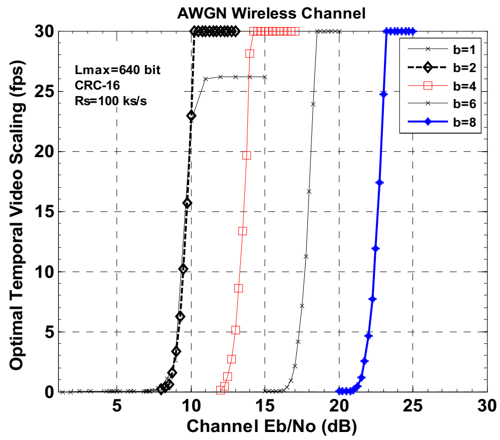

Fig. 6 Optimal video quality vs. received SNR in AWGN channel based only SR-ARQ scheme with variable modulation scaling (b).

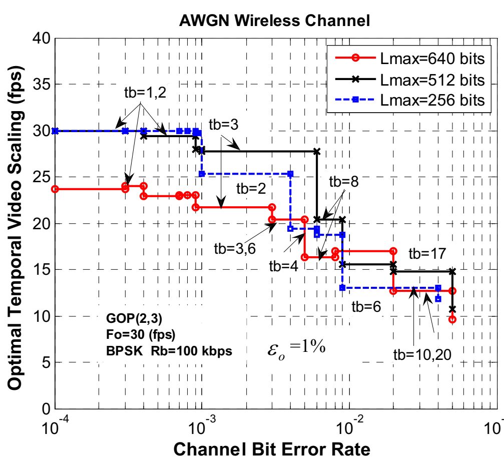

Fig. 7 Optimal video quality of the adaptive link mode based FEC code for various packet length values.

Table III shows adaptive FEC scheme under various values of system threshold of residual packet loss (error) rate. It is found that only optimal t_b of 1 or 2 will generate a good video quality when ε_o increases between 1% and 3%. As a result, we can conclude that as ε_o increases, the system performance dramatically degrades at the client end. In addition, as FEC code increases (i.e., optimal t_b increases), there exists no significant improvement in the perceptual video quality, but the cost of power consumption due to the FEC overhead will rise. Therefore, in our model the video system settings must not exceed 3% for predefined threshold and then the only optimal t_b and modulation scaling b will be adapted to the channel state information.

On the other hand, the proposed cross-layer model based on adaptive RS FEC and chosen modulation format can also improve the MPEG-4 system quality performance compared with other studies. In [8], the video quality is based only BCH FEC scheme for BPSK modulation to provide approximately 22-26 [fps] with packet length of 511 bits at SNR/bit of 7.35-8.4 [dB]. Moreover, Lo *et. al.* [4] applied integral tool environment based MOS-AM for RTP/RTCP/UDP/IP stack to evaluate 25 [fps] at MOS Score 5 (Excellent) and PSNR of 37 [dB] using OPNET without taking the impact of modulation format on RLC packet at the data link layer.

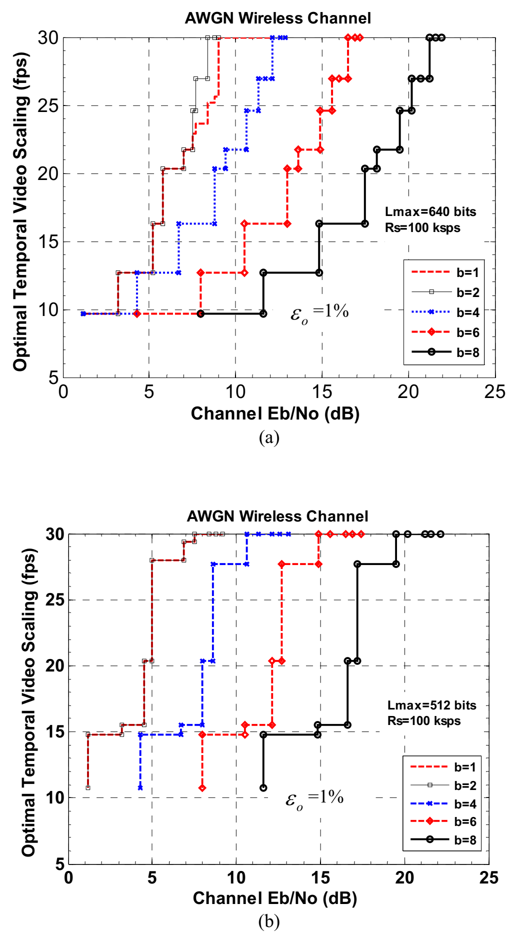

Fig. 8 Optimal video quality of the adaptive link mode based FEC code for various values of modulation scaling (b). (a) L_{max} =640, and (b) L_{max} =512.

VI. Conclusion

In this paper, we have proposed an analytical adaptive link model to improve the link reliability via dynamically adapting a TCP throughput for MPEG-4 video traffic over a UMTS network. The adaptive approach has been based on heuristic TCP function, forward error correction (FEC), and modulation format. The perceived video quality can be dynamically optimized by providing the optimal allocation bandwidth required when the optimal code efficiency of byte-level RS FEC and/or CRC SR-ARQ is achieved with the appropriate modulation scaling under different channel state estimations.

On the other hand, the proposed scheme can be modified for the higher video source rate (i.e., symbol rate) [29], for example 10 Mbps, to support the multimedia applications of HSDPA for UMTS network [1]. Future work can involve FSMC (Markov system model) and optimize the selection of modulation levels and packet sizes over a correlated Rayleigh fading channel in order to provide more accuracy in the video quality transport.

TABLE II
IMPACT OF MODULATION SCALING FOR ONLY CRC *SR*-ARQ SCHEME

b	L_{max}	SNR/bit [dB]	P_{LOSS} %	PFR [fps]
1	640	10 (BER=$3x10^{-6}$)	0.19	24
2	640	10	0.25	23
4	640	13.85	0.25	24
6	640	18.25	0.23	24
8	640	23	0.22	24

TABLE III
IMPACT OF SYSTEM RESIDUAL PACKET ERROR FOR ADAPTIVE FEC SCHEME

b	*RS* code	opt.t_b (byte)	SNR/bit [dB]	ε_o	PFR [fps]
1,2	(80,78)	1	9.0 ($1x10^{-5}$)	1 or 3%	25.59
1,2	(80,78)	1	8.4	1%	23.67
4	(80,78)	1	12	1%	24
6	(80,78)	1	16.5	1%	24
8	(80,78)	1	21.2	1%	24
b	*RS* code	opt.t_b (byte)	SNR/bit [dB]	ε_o	PFR [fps]
1,2	(80,76)	2	7.5 ($4x10^{-4}$)	2%	22.9
4	(80,76)	2	11.4	2%	22.9
6	(80,76)	2	15.6	2%	22.9
8	(80,78)	2	20.2	2%	22.9

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