Performance Evaluation of MPEG-4 Video Transport in Rayleigh Fading Channel

Ghaida A. AL-Suhail
Department of Computer Engineering, University of Basrah, Basrah, Iraq.
Email: gaida_alsuhail@yahoo.com.

***Abstract* — In this paper, we present a new analysis of the optimal throughput of a single user for transporting MPEG-4 video over Universal Mobile Telecommunication System (UMTS) network. A wireless channel considers that bit errors associated with link-layer packets are due to the time-varying features of a Wideband Code-Division-Multiple Access (WCDMA) channel. By this network, the perceived video quality can be evaluated based on the heuristic function of TCP throughput which is dependent of the user's mobility (Doppler frequency) over a Rayleigh fading channel. The results show that a video quality of service (QoS) in terms of temporal scaling can be significantly adapted at fast fading region when the maximum network throughput of acknowledged mode is achieved under optimal packet length in radio-link layer.**

Index Terms— MPEG-4 Video, UMTS, WCDMA, TCP, Video Quality.

I. INTRODUCTION

Universal Mobile Telecommunications System (UMTS) is a third-generation (3G) cellular network that enables multimedia applications access via the radio interface using Wideband Code Division Multiple Access (WCDMA) [1-2]. It can provide maximum data-rates ranging from 64kbps to 2Mbps in different environment types. Recently, these networks are used widely in many commercial services such as a wireless Internet, mobile computing and cellular telephoning. For instance, it 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. Also, 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. However, such wireless communication channels are prone to errors due to various physical impairments. Therefore, the key issue in this paper is to evaluate the video quality performance over error-prone UMTS network.

Fortunately, a video streaming is nowadays a promising multimedia application and a key factor to success of 3G systems. In these applications, mobile users can access online video clips (such as news, sports, etc.) by clicking on a hyperlink using their web browser. Then, a video player starts playing with the selected clip. With steaming, the video content need not be completely downloaded, but the client can begin playback the video a few seconds after it begins receiving parts of the content from the video server. Since raw video requires high transmission bit rates, video compression is usually employed to achieve transmission efficiency. MPEG-4 [3], therefore, is a video compression standard adopted by most mobile networks including UMTS. This compression standard provides good video quality at bit rates that are suitable for mobile and wireless transmission.

However, many popular wireless multimedia networks cannot provide a guaranteed quality of service (QoS) in spite of the increase in demand on multimedia applications such as real-time video streaming, video conference, and video on demand. To this end, it is essential to rely on QoS metrics pertinent to wireless links in terms of data loss, delay, and throughput. In practice, many major challenges of video traffic are faced on the wired and wireless Internet links [1-4]. Some of these challenges deal with high packet loss rate due to the congestion of buffer overflow over wired networks; and others are mainly faced by the characteristic of wireless links, which are mostly suffering from low bandwidth and high bit error rates due to the noise, interference, unpredictable user mobility (Doppler frequency), multi-path fading and time-dispersive effects introduced by the wireless air interface. In fact, these 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. These transmission rate decreases are unnecessary and lead to resource inefficiency. Therefore, there are two types of solutions to solve this problem: modifying TCP model and improving the link reliability observed by TCP [4-7]. The later solution may include adaptive rate control [8], adaptive selective Repeat (ASR) protocols (re-transmission) [9], adaptive modulation [10], forward- error-correction (FEC) at packet-level [11-13] and/or channel bit-level [14].

In this paper, we propose a new analysis to improve the link reliability via providing a maximum TCP throughput for MPEG-4 video traffic over a UMTS network. First, a wireless channel is accessed via TCP throughput with heuristic function to explain the effect of user's mobility in term of the Doppler frequency over a Rayleigh fading channel. A Cyclic Redundancy Code (CRC) channel coding at radio link layer is employed. Thus the physical layer can capture a Signal-to-Noise Ratio (SNR) versus bit error rate (BER) through a simple Binary-Phase-Shift-Keying (BPSK) modulation in case of Rayleigh fading channel with some restriction on the packet length. Second, to maximize the network throughput and to enhance the perceptual video quality, optimal packet length can be achieved according to the channel state estimation. As a result, the proposed analysis can

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predict MPEG-4 video quality performance by adapting a playable frame rate (temporal scaling) under various Doppler frequency conditions.

The rest of the paper is organized as follows. Section II presents the system model followed by Section III for our proposed model. In Section IV, we derive the analytical optimal temporal model for MPEG-4 video. Simulation results are explained in Section V, and finally Section VI summarizes conclusions.

II. SYSTEM MODEL

A. MPEG Video

We consider MPEG-4 video stream in our system model. MPEG-4 is a set of tools to create, represent and distribute individual audiovisual objects, both natural and synthetic, ranging from arbitrarily shaped objects to sprites, face and body animations [15]. In general, MPEG defines three picture types in terms of temporal processing. They are I (intra-frame coded) pictures, P (forward temporal predictive coded) pictures, and B (bi-directional temporal predictive coded) pictures. In MPEG video coding, pictures of an input video sequence are often divided into groups of pictures (GoP), each GoP may contain one I picture, a number of P pictures, and optionally a few B pictures between I or P pictures. The structure of a GoP can often be described by two variables: 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 $N_B = (1 + N_P) \times N_{BP}$ [11]. Figure 1 shows an example of a GoP with N_P=2 and N_{BP}=2, where the subscripts are the pictures temporal references in display order [16].

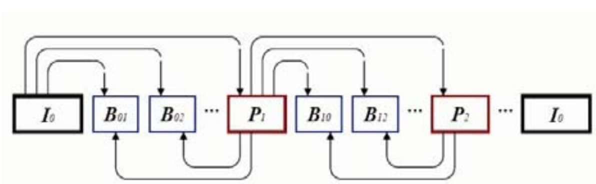


Fig. 1 Example of GoP structure with GOP(2,2).

B. UMTS

Network architecture of the UMTS is designed for only packet-switched operation as shown in Fig. 2. It consists of one or several User Equipments (UEs), the UMTS Terrestrial Radio Access Network (UTRAN) and the core network. UTRAN has two interfaces, one is with UE using WCDMA, and the other one is with core network. The UTRAN is composed of Node Bs (Base Stations) connected to a Radio Network Controller (RNC). The core network, which is the backbone of UMTS, comprises the Serving GPRS Support Node (SGSN) and the Gateway GPRS Support Node (GGSN). The SGSNs route packets to and from UTRAN, while GGSN interface with external IP networks. UE, which is a mobile station, is connected to Node B over the UMTS radio interface.

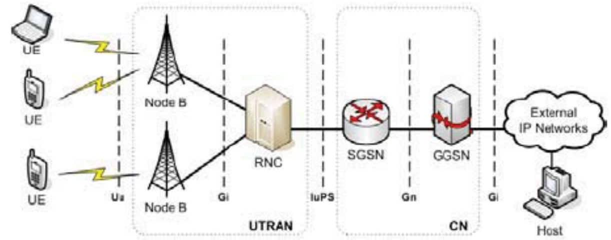


Fig. 2 A typical UMTS architecture network

Since the UMTS protocol architecture is based-IP for the transmission of user data, the applications and the Internet Protocol are located at the end-nodes, namely, the UE and a host. In the link-layer, the functionality of the Radio Link Control (RLC) layer can operate in three different modes: *acknowledged*, *unacknowledged*, and *transparent*. The acknowledged mode (RLC/AM) provides reliable data transfer over error-prone radio interface such as TCP based ones. In contrary, both the unacknowledged and transparent modes do not guarantee data delivery. The transparent mode is designed for the UMTS circuit-switched mode in which data are passed through the RLC unchanged [1,9]. In this paper, we focus only on RLC/AM because this protocol performs error detection and recovery for applications requiring a reliable channel.

C. Network Model

To simplify the function of RLC (hardware-radio layer), Fig. 3 illustrates a typical point-to-point wireless network model between Base station transmitter and a single mobile receiver in UTRAN for video transporting. In this system model, we focus on the point-to-point QoS mainly for the following reasons [4]. First, this model provides fundamental studies for end-to-end QoS guarantees in terms of throughput and bit and/or packet loss rate. Second, the bottleneck of most current wireless networks (e.g., cellular networks) is typically located at the last wireless hop, which can be simplified and modelled as a point-to-point QoS provisioning problem.

Upon above, various errors encountered over a link can mainly be classified as packet loss due to overflow buffer (congestion) and/or error bits due to wireless features environment including additive White Gaussian noise, interference, fast-fading, etc. [17-19]. Also, channel state information (CSI) can be estimated at the receiver and fed back to the transmitter for selecting adaptive modulation and/or channel coding such as CRC and RCPC, and RS codes. 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.

Within this model, video input needs to be encoded by codec 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. However, the error probability and the user's mobility are essential parameters for a perceptual video quality at the client end.

For more robust model, we assume Acknowledged Mode (AM) to guarantee the packet delivery by retransmission 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. In this work, we consider SR-ARQ scheme where the RLC blocks retransmitted are those that receive a negative acknowledgement.

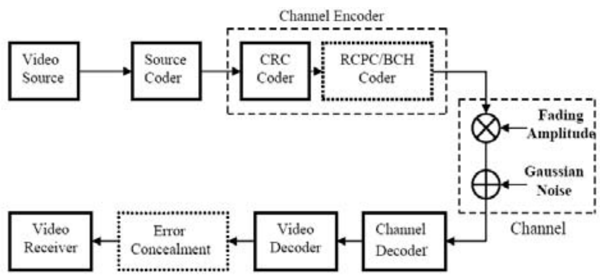


Fig. 2 A typical wireless point-to-point video communication system corrupted by fading and AWGN noise.

D. *Fading Channels*

The users in WCDMA systems [6] are usually sending messages at the same time and with the same bandwidth but with different codes. The main noise source of each user is the interference signals from other users due to the imperfect of orthogonal spreading codes and the channel dispersion characteristics such as multi-path and Doppler shift. These problems limit the capacity of the mobile communication system. To increase the wireless capacity (bandwidth access) and to enhance the channel performance, one should optimize the available radio resources. In this paper, we consider the channel is corrupted by unpredicted user mobility as well as additive White Gaussian noise. The improving of the link reliability by TCP can be achieved via a maximum achievable throughput at optimal packet length when CRC redundancy code is employed via retransmission scheme.

Specifically, the fading characteristics of different radio channels and their associated effect on communication performance have been studied extensively. Despite the fact that Rayleigh fading is the most popular model, Rician fading is observed in mobile radio channels as well as in indoor cordless telecommunication systems [1]. In cellular system, Rayleigh fading is often a feature of large cells, while for cells of smaller diameter, the envelope fluctuations of a received signal are observed to be close to Rician fading. A slow and flat Rician fading model means that the duration of a symbol waveform is sufficiently short, then the fading variations cause negligible loss of coherence within the received symbol [19].

E. *Error Control*

In Fig. 2, we consider a realistic video transmission system which consists of a transmitter, a receiver, and a communication channel with a limited bandwidth B_w. The transmitter constructs packets of K bits and transmits the packets in a continuous stream. To ensure that bits received in error are detected, the transmitter attaches a C bit CRC FEC to each data packet, making the total packet length $K + C = L$ bits. This packet is then transmitted through the air and processed by the receiver. The FEC decoder at the receiver is assumed to be able to detect all the errors in the received packets. (In practical some errors are not decodable, but this probability is small for reasonable value of C and reasonable SNRs).

Upon decoding the packet, the receiver sends an acknowledged, either positive (ACK) or negative (ACK), back to transmitter via ARQ scheme [9]. 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.

More precisely, the source coder provides compression (usually lossy) of the video while the channel coder introduces redundancy in order to combat error caused by a noisy channel. The concealment stage is a post-processing stage (usually found only in lossy compression systems such as video) which is useful for reducing the effects of residual channel errors. In this stage, operations such as spatial or temporal filtering are carried out to improve the quality of corrupted video. For sake of simplicity, the concealment stage and the channel coding (e.g. RCPC, BCH, or RS) are not considered in our proposed approach. Thus we assume a typical model of wireless video communication; whereby a video server sends a video stream to a receiver via a wireless channel corrupted highly with fast-fading, and no interference from other signals.

III. Proposed Approach

In this section, we deal with the effect of Doppler frequency on the video transporting over time varying channel. We, therefore, consider the wireless link is characterized by a limited bandwidth, i.e. B_w. Further, the effective packet loss rate P_{Loss} is only arising due to the corruption of bit errors ignoring the congestion of opening many concurrent TCP video connections on the same channel. Then, we can use the following TCP model to analyze the problem.

A. *Proposed Model*

As a fact, throughput of a system is generally defined as the number of payload bits per second received correctly. At hardware-radio link layer (physical layer), we therefore consider $\Re_b$, with $\Re_b \le B_w$, be the available bit rate (bandwidth) for a given TCP connection. The throughput can be roughly represented at the hardware-link layer as follows [21],

$$B_{Phy,m} = \Re_C \cdot \Re_b \cdot f(\gamma_b, L) \qquad (1)$$

We will refer to the term "*mod m*" to indicate to a specific choice of an uncoded modulation. $B_{Phy,m}$ represents a maximum channel throughput as the number of payload bits per second received correctly for uncoded BPSK scheme [18], and $\Re_C$ is a redundancy ratio,

$$\Re_C = \frac{L-C}{L} = 1 - \frac{C}{L} \tag{2}$$

L is a packet length where each packet is transmitted bit by bit through the channel. The channel is assumed narrowband (flat-fading), so the power spectra of both the received signal and the noise have no frequency dependence, i.e. the channel is characterized by a single path gain variable.

Moreover, $f(\gamma_b, L)$ denotes an appropriate function, which may describe link errors by providing the impact of the channel signal-to-noise ratio per bit and the packet length on TCP throughput performance [21]. However, other definition can be found in [7] 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_b is the SNR per bit given by

$$\gamma_b = E_b / N_o = \frac{P}{N_o \Re_b} \tag{3}$$

For uncoded BPSK scheme, E_b, N_o, and P represent the bit energy, the one-sided noise power spectral density, and the received power respectively. As one consequence, the function $f(.)$ can be expressed as [7]

$$f(\gamma_b, L) = [1 - P_e(\gamma_b)]^L \tag{4}$$

and consequently (4) can be rewritten as,

$$P_{Loss}(\gamma_b, L) \leq 1 - (1 - P_e(\gamma_b))^L \tag{5}$$

L denotes a packet length (in bits). P_{Loss} is a packet loss rate due to link bit errors. Thus we can define $P_{Loss}(L, \gamma_b)$ as the probability of packet error in terms of packet length in L bits and γ_b which is being SNR per bit for uncoded modulation scheme. It is noticed in (1) that at high SNR, i.e., high γ_b, $f(\gamma_b, L) \approx 1$ and the throughput of physical layer will proportional to $1 - C/L$. Therefore, the throughput gain becomes negligible if we increase L beyond a certain point [21].

B. Optimal Packet Length

In this section, we handle the analytical solution for the optimal packet length L^* by assuming L has continuous values. By differentiating (1) with respect to L and using the formula of packet success rate (PSR) function in (4), we produce

$$\frac{\partial B_{Phy,m}}{\partial L} = \frac{C}{L^2} \Re_b f(\gamma_b, L) + \left(1 - \frac{C}{L}\right) \Re_b f(\gamma_b, L) \ln(1 - P_e(\gamma_b)) \tag{6}$$

Setting (6) to zero and solving the resulting quadratic equation as in [21], then L^* yields

$$L^*(\gamma_b) = \frac{C}{2} + \frac{1}{2}\sqrt{C^2 - \frac{4C}{\ln(1 - P_e(\gamma_b))}} \tag{7}$$

C. Analytical TCP Throughput Prediction

Several formulas have been pursued to approximate the behavior of TCP throughput, e.g. [5-7] at the hardware-link layer. In this paper, we use the heuristic formula proposed in [5] which is the most accurate for wireless communication channels. In order to predict the TCP throughput, the observed shape of its curves tends to be a constant by equaling one minus the percentage of overhead for $P_{Loss} \to 0$ and zero for $P_{Loss} \to 1$. However, one suggestion is a numerical fit to involve the logarithm of P_{Loss}. In this formula, the shape of TCP transition can depend on the value of the Doppler frequency. Hence, a TCP throughput model can be defined through a heuristic function $\wp(x)$ as [6],

$$\wp(\xi) = S(0) \cdot \begin{cases} \dfrac{10^{\alpha \ln\left(\frac{1}{\xi - \xi_s} - 1\right)}}{10^{\alpha \ln\left(\frac{1}{\xi - \xi_s} - 1\right)} + 1} & \text{if} \quad \xi > \xi_s \\ 1 & \text{if} \quad \xi \leq \xi_s \end{cases} \tag{8}$$

where,

$$\xi = 1 + \left[\log_{10}(P_{Loss})/3\right] \tag{9}$$

$$\xi_s = \frac{1}{(\theta . f_d + \beta)} - \kappa \tag{10}$$

$\wp(x)$ function is independent of the network load (number of users) and is parameterized only by the Doppler frequency. Now, we let the achievable upper-bound of the network throughput of a TCP connection is defined by,

$$B_{TCP} \cong \wp(x) \tag{11}$$

Note that f_d in (8) is the Doppler frequency in Hz used to model the Rayleigh fading, and $S(0)$ is the average TCP throughput for error-free link, i.e., $P_{Loss} = 0$. Meanwhile, the constants θ, α, β, and κ are defined as θ=1.39, $\alpha = 1.3$, β=2.78, and κ=0.03. A typical value of f_d is f_d=6 Hz [6]. In fact, these involved constants do not have specific physical meaning, but they satisfy the condition of giving an analytical expression for throughput which is reasonably close to actual points obtained by fitting the simulation curves.

In case of error-event $P_{Loss} \to 1$, if the available bit rate $\Re_b$ is being the effective average TCP throughput in (8), then by replacing the product term $\Re_b \cdot f(\gamma_b, L)$ in (1) by B_{TCP} we produce another heuristic expression for the predicted TCP throughput in the presence of redundancy and it is only dependent of Doppler frequency. Thus (1) can be rewritten in another form as,

$$B_{Phy,m} = \Re_C \times B_{TCP} \tag{12}$$

Equation (12) is being subjected by a network bandwidth constraint that $B_{Phy,m} \le B_w$ and $B_{TCP} \le \Re_b < B_w$.

D. Rayleigh Fading Channel

In fast fading channels where the channel gain changes within a bit period, we need to use the average error probability instead of (5), and PSR in (4) also needs to be modified accordingly. So, we treat our modification in (11) and (12). For BPSK, the average probability of bit error in a flat Rayleigh fading channel is approximately given by [18]

$$\bar{P}_e(\gamma_b) = 0.5\left(1 - \sqrt{\bar{\gamma}_b \Big/ \left(1 + \bar{\gamma}_b\right)}\right) \tag{13}$$

and PSR becomes,

$$g(\bar{\gamma}_b, L) = \bar{P}_{Loss} = [1 - \bar{P}_e(\bar{\gamma}_b)]^L \tag{14}$$

In Rayleigh fading channel, to achieve optimal TCP throughput, the optimal packet length $L^*(\bar{\gamma}_b)$ can be again obtained by substituting (14) into (8).

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 over AWGN channel can be applied for upload/download streams. Since p_b in AWGN channel decays exponentially as γ_b increases, the probability of bit error can be given by [20],

$$P_e(\gamma_b) = Q\left(\sqrt{2\ \gamma_b}\right), \tag{15}$$

$Q(.)$ is Gaussian cumulative distribution function. Fig. 4, shows optimal packet length as a function of wireless SNR per bit for BPSK. In AWGN channel, L^* grows rapidly as SNR increases. In practice, this increase may cause problems because larger packet size means higher delay and more memory requirement. By using (1)-(7), we can obtain the optimizing throughput system under a condition of optimal packet length. In this case, the product $\Re_b \cdot f\left(\gamma_b, L^*\right)$ must be adapted to keep γ_b at high values in dBs to achieve the optimal throughput [21]. In Rayleigh fading channel, L^* is expected much smaller and asymptotically proportional to square root of the average received SNR.

IV. Optimal Temporal Model

We take the following scenario to evaluate video quality performance:

- A flow is considered with data rate (throughput) not exceeding the maximum data rate of TCP connection in the same network conditions. Here, the TCP sending rate is controlled in accordance with network conditions as TCP does, on the wired Internet [8]. By adjusting the sending rate to the desirable rate determined by an underlying TCP, one can achieve the required video quality of video applications over a wireless link.
- There is no extra-traffic due to concurrent TCP video connections on wireless channel. The wireless bandwidth link is limited and there is no congestion of video connections. The packet loss is only due to wireless channel bit errors. In contrary, the effect of any round-trip time due to ARQ scheme is ignored in this analysis. The only restriction is wireless channel capacity which should imply $B_{TCP} \le B_w$. The backward route from video receiver to video server is assumed to be congestion-free but not error-free due to bit errors [8].
- Optimal control rate should result in the highest possible throughput and the lowest packet loss rate by using (5). To avoid any network instability, $\Re_b$ is regarded as the available bandwidth for video streaming and by adjusting the video traffic, the high-quality video play-out at a receiver can be expected. Hence, for under-utilized channel, $B_{TCP} \le \Re_b < B_w$ holds when only one TCP connection exists for single user.

Using this scenario, the optimal throughput in (12) can be again expressed under optimal packet length for various error-conditions over Rayleigh fading channel.

In order to achieve maximum performance in an erroneous noisy channel environment, a careful design of the channel coding is important. Thus CRC 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 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 [9]. In case of CRC code, the maximum achievable TCP throughput can be expressed as in the presence of redundancy ratio,

$$B^*_{Phy,m} = \frac{L^* - C - 16}{L^*} \times B^*_{TCP} \tag{16}$$

where B^*_{TCP} denotes the maximum TCP throughput introduced due to Doppler frequency effect (generated by user mobility) over Rayleigh fading without taking into account the CRC redundancy code. Hence, the resultant playable frame rate (PFR) R can be computed as [11]

$$R^*_{eff} = G.W_I \left[1 + \chi_P + N_{BP} \cdot W_B \cdot \left(\chi_P + W_I \cdot W_P^{N_P} \right) \right] \quad (17)$$

where,

$$\chi_P = \frac{W_P - W_P^{N_P+1}}{1 - W_P}, \qquad W_i = (1 - p_w)^{S_i}, \quad \text{and}$$

$$G^* = \frac{B^*_{Phy,m} / L^*}{S_I + N_P S_P + N_B S_B}. \quad (18)$$

W_i stands for the successful transmission probability of the *i-th* frame type (I, P, and B) in a GoP pattern without taking into account any packet FEC correction at application layer, and S_i denotes packet size of the *i-th* frame type. In our analysis, optimal packet length (L^*) must not exceed 640 bits over UMTS network. $B^*_{Phy,m}$ is the effective network throughput received at the client in (bps), G^* corresponds to 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).

V. Simulation Results

Table I describes simulation parameters setting used for MPEG-4 video stream over WCDMA wireless network [2,11]. A channel capacity is assumed at the limited bandwidth B_w in order to be compatible to 1xRTT CDMA network, and the upper-bound of the network throughput therefore does not exceed B_w. Moreover, the UMTS PHY model considers transport blocks (packets) are not sent over Dedicated Physical Channel (DPCH). In practical, DPCH is mainly used to maintain a fixed bit rate for the duration of the connection. For simplicity in simulation, the effect of RLC packet header (16 bit) in (16) is ignored. The video quality can be subsequently dependent of a variable TCP throughput under fast fading channel conditions. Our simulation results in Figures 4-6 are based on MatLab programming. They illustrate clearly the performance evaluation of MPEG-4 video stream over UMTS network under different channel conditions.

Fig. 4 depicts a comparison between AWGN and fast Rayleigh fading channel to provide optimal packet length and its corresponding packet loss rate. It is clearly noticed that AWGN channel introduces a greater optimal packet length for video transmission than Rayleigh channel. In AWGN channel, the maximum packet length would be rapidly 640 bit in our simulation at channel SNR region greater than 9 dB [21].

In contrary, the fast fading channel in Fig. 4 introduces gradually exponential increase in the optimal packet length as channel SNR increases accordingly. This will obviously reflect on TCP network bandwidth when the speed of mobile user is changed between 2Hz - 80 Hz, and it becomes independent (i.e., close to i.i.d case) when Doppler frequency closes to high

TABLE I
Simulation Parameters for MPEG-4 video stream

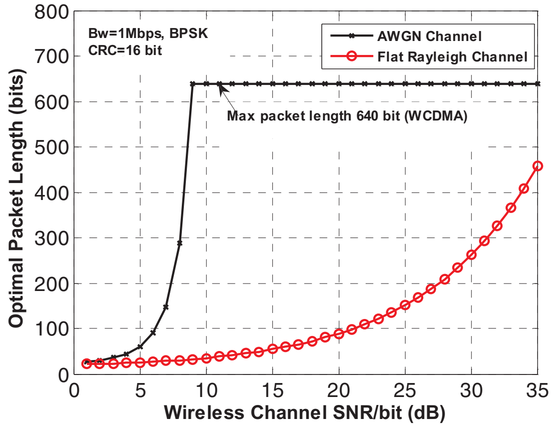
WCDMA Network	
Channel Capacity B_w	1Mbps for 1xRTT CDMA
Carrier frequency	2.4 GHz
Max Packet length (L_{max})	640 [bit] for WCDMA
Physical Channel type	γ_b (AWGN), $\bar{\gamma}_b$ (Rayleigh) Channel SNR/bit
	Modulation Scheme BPSK (upload/download)
	Not required for DPCH
	Max. Download bit rate 384 [kbps]
	CRC 16 [bit]
	Transport Packet Loss 0-25%
TCP Throughput Parameters	θ=1.39 , $\alpha = 1.3$, β=2.78 , κ=0.03
Application Layer	**MPEG-4**
GOP(2,3) 12 frames	IBBBPBBBPBBB
Video source	30 [fps]
S_I, S_P, S_B	25 [packet], 8 [packet], 3 [packet]

values greater than 80Hz (see Fig. 5). This observation indicates that as the Doppler frequency increases the performance degrades, i.e., slower channels correspond to better performance as observed in [6]. Within the scenario in Section 4, the optimal TCP throughput must be limited up to 384 kbps to achieve the upper-bound of channel capacity constraint in UMTS network. Therefore, Fig. 6 depicts the resultant video QoS in terms of temporal quality (i.e., number of resultant frames per second) under different Doppler frequency conditions. It is found that a full video motion at 30 [fps] can be achieved at channel SNR equal or greater than 30 [dB] when fast fading is dominant. Meanwhile, a one-third of original play-out rate (i.e., only 10 [fps] for instance) can be achieved for channel SNR values of 25-30 [dB]. However, it is clearly shown that the optimal transport packet loss rate (PLR) is roughly 6% to 12% although the CRC scheme is used at the radio-data link layer. It can be concluded that video quality decline rapidly when PLR increases because the errors will propagate through frames.

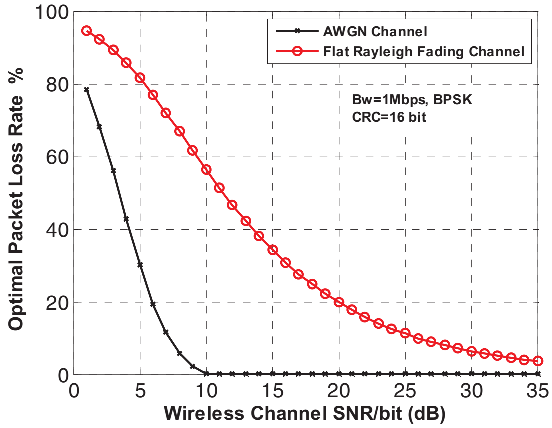
As observed in Table II, our approach explains a good predicted video quality performance as compared with other studies. For instance, video quality in terms of Mean Opinion Score (MOS) and video frame rate error were evaluated by Lo *et al.* [2] for the perceptible MPEG-4 video over UMTS network using integrated tool environment. It is noticed that our proposed analytical approach based heuristic TCP throughput analysis outperforms the approach based UDP in [2] when Doppler frequency effect increases. Meanwhile, in Table III, we introduce a video quality over WCDMA channels under different error-conditions and user mobility compared to hybrid scheme of BCH FEC and ARQ in [13]. Within our analysis, a high channel SNR must be achieved to adapt a fast fading region under optimal packet length of 260 or 261 bits.

VI. Conclusions

The paper has evaluated the performance of transporting MPEG-4 video over UMTS networks using a new analytical framework to provide the perceived video quality. The optimal video quality is evaluated in terms of temporal scaling (frames per second) of a single user based on a heuristic TCP throughput function over a fast Rayleigh fading WCDMA channel. The analysis applies CRC channel coding at the radio

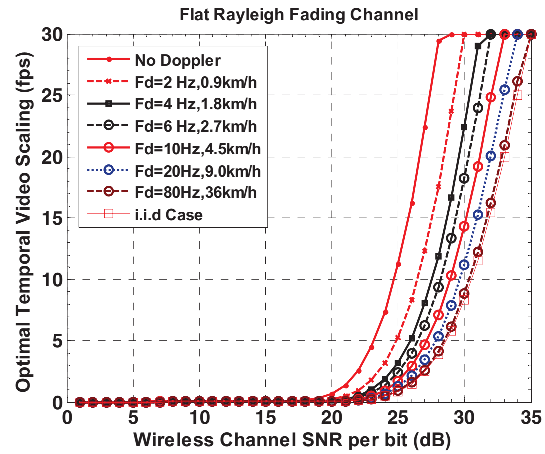


(a)

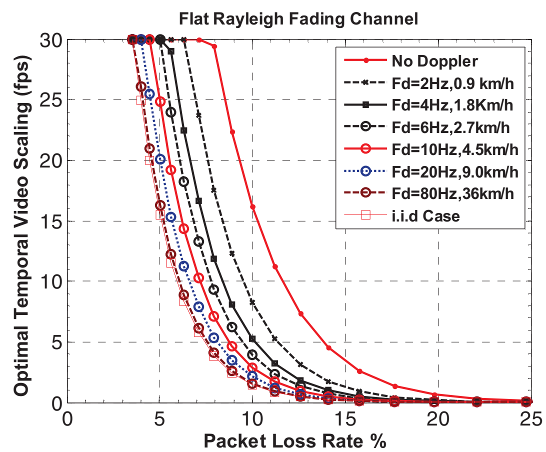


(b)

Fig. 4 Performance evaluation vs. channel SNR/bit. (a) Optimal packet length, (b) Optimal packet loss rate.



(a)



(b)

Fig. 6 Optimal video quality under various Doppler frequency values vs.
(a) wireless channel SNR/bit using BPSK scheme, and
(b) corresponding optimal packet loss rate.

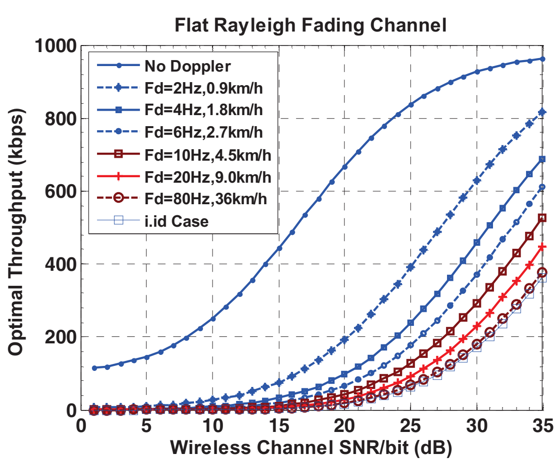


Fig. 5 Optimal TCP heuristic throughput function vs. channel SNR/bit over only Rayleigh fading channel.

link layer to improve the bandwidth access from the wireless link when a user's mobility (Doppler frequency) is considered. Furthermore, the video traffic is assumed to be controlled by TCP and Acknowledged Mode (AM) of Automatic Repeat Request (ARQ). Simulation results show that the perceived video quality achieved is superior when optimal packet length is achieved. In other words, video quality at the client can be significantly adapted at fast fading region when a maximum network throughput is achieved under optimal packet length. Further work can be extended to involve channel coding and adaptive modulation for more robust wireless video transport.

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TABLE II
A COMPARATIVE EXAMPLE FOR MPEG-4 VIDEO STREAMING OVER UMTS NETWORK

Approach	Transport Protocol	Error Type	Video Quality (or PFR)	Average PLR%
Integrated Tool Environment Based MOS-AM [2]	UDP	Radio Interface (No congestion or bit errors)	MOS Score 5 (Excellent) (37 dB PSNR, 25 fps) MOS Score 3 (Fair) 12 [fps]	0-0.3% 35%
Proposed Approach Based -AM	TCP	Fast fading channel (Doppler frequency) 0-80 Hz	25 [fps] 12 [fps]	4%-9% 6%-11%

* MOS-AM=Mean Opinion Score -Acknowledged Mode

TABLE III
PERFORMANCE EVALUATION OF MPEG VIDEO STREAM WHEN A MAXIMUM THROUGHPUT IS ACHIEVED UNDER VARIOUS CHANNEL CONDITIONS

Approach	Channel State	Error Type	BCH Parity bits	Packet length (bits)	Optimal Bandwidth (kbps)	*PFR* (fps)
Hybrid FEC and ARQ-Based TFRC [13]	C1 (8.40 dB)	1×10^{-4} Random Error AWGN	9 (t=1)	511	80.11*	26.13
	C2 (7.35 dB)	5×10^{-4} Random Error AWGN	18 (t=2)	511	71.08	22.13
Proposed Approach Based Heuristic TCP function (16 bit CRC- AM)	C3 (29.3dB)	WCDMA error= 2.93×10^{-4}, f_d =2 Hz, speed=0.9km/h	0 (No FEC)	260 (L^*)	384*	26
	C4 (29.8 dB)	WCDMA error= 2.6×10^{-4}, f_d =4 Hz, speed=1.8km/h	0 (No FEC)	261 (L^*)	384*	22

* Hybrid FEC and ARQ: Upper-bound bandwidth (network throughput) achievable is 80.11 kbps, Fixed RTT=168 ms, GoP(2,3)
* Proposed Approach: Upper-bound bandwidth (network throughput) achievable is 384 kbps, GoP(2,3)

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