تأثير حجم الرزمة على المواصفات المؤقتة لنقل صور فيديو لا سلكياً

الحارث عبد الكريم الخفاجي

كلية العلوم – جامعة بابل

الخلاصة : في هذا البحث تم تطبيق نموذج تقليل رزمة بشكل تحليلي المجموعة صور (Gops) نموذج (mpeg4) من خلال شبكة الانترنيت اللاسلكي ، افترض أن Awgn قناة لاسلكية تسبب أخطاء متتابعة صغيرة تتزامن مع الرزم ، لذلك تاثير حجم الرزمة هو أول ما تم اختباره على مواصفات نوعية الفيديو في حالة نقل معدل سيطرة ودي TCP ومن ثم مقارنة نوعية الفيديو في حالة عدم نقل TFRC اظهرت النتائج الجديدة .. المفترضة أنه يمكن تقييم أداء مواصفات نوعية الفيديو والمتوقع أن تكون جيدة من حيث نسبة الصورة القابلة للمعاملة (صورة / ثانية) عند مستويات مختلفة لقناة أحادية إلى نسب الضوضاء في (dBs) عندما يختار حجم رزمة مناسب قبل بدء الإرسال فضلاً عن مشابهة نوعية الفيديو الإدراكية إلى واحدة من شبكات الأنترنيت اللاسلكية .

References:

1. F. Zahi, Optimal Cross-Layer Resource Allocation for Real-Time Video Transmission over Packet Lossy Networks, Ph.D. thesis, Dept. of Elect. And Comp. Eng., North Western University, June 2004.

2. T. Wang H. Fang, and L. Chen, "Low-Delay and Error-Robust Wireless Video Transmission for Video Communications”, IEEE Trans. on Circ. And Sys. for Video Tech., vol. 12, no. 12, pp. 1049-1058.Dec.2002. om

3. W. S. Lee, M. R. Pickering, M. R. Frater, and John F. Arnold, "A Robust Codec for Transmission of Very Low Bit-Rate Video over Channels with Bursty Errors," IEEE Trans. on Circ. and Sys. for Video Tech., vol. 10, no. 8, pp. 1403-H12, Dec. 2000.

4. F, Yang, Q. Zhang, W. Zhu, and Y.Q. Zhang, "End-to-End TCP- Friendly Streaming Protocol and Bit Allocation for Scalable Video over Wireless Internet", IEEE Journal on Selec. Area In Comm., vol. 22, no. 4, May 2004.

5. M. Chen and A. Zakhor, “Multiple TFRC Connections Based Rate Control Wireless Networks", IEEE Trans. On Multimedia, vol. 8, no. 5, Oct. 2006.

6. N. Feamster and H. Balakrishnan, "Packet Loss Recovery for Streaming Video", part of Master thesis, Dept. of Elec. Eng. and Comp. Sc., Massachusetts Institute of Technology, May 2001.

7. H. Wu, M. Claypool, and R. Kinicki, "Adjusting Forward Error Correction with Quality Scaling for Streaming MPEG," NOSSDAV '05, pp. 111-116, June 2005.

8. Y. Yuan, "A GoP Based FEC Technique for Packet Based Video Streaming," Proc. of the 10th WSEAS Inter.

Conference on Comm.. Greece, pp. 187-192, July 10-12, 2006.

9. G. A. AL-Suhail and L. Tan, "Improving the QoS of Wireless Video Transmission via Packet-level FEC," Proc. of Inter. Conf. on Signal Processing and Communication Systems (ICSPCS'2007), Gold Coast, Australia, 17-19 Dec 2007.

10. Y. Pei and J. W. Modestino, "Interactive Video Coding and Transmission over Heterogeneous Wired-to-Wireless IP Networks using an Edge Proxy," EURASIP J. on App. Sig. Proc., pp. 253-264,2004.

11. D. Krishnaswamy and M. Schar, "Adaptive modulated Scalable video Transmission over Wireless Networks with Game Theoretic Approach," IEEE Multimedia Signal Processing Workshop, MMSP'04, 2004.

12. S. Khan, M. Sgroi, E. Steinbach, and W. Kellerer, "Cross-Layer Optimization for Wireless Video Streaming-Performance and Cost," IEEE Conf. on Multimedia & Expo, ICME '05, Amsterdam, July 6-8, 2005.

13.X. Tong, W. Gao, Q. Huang, "A Novel Rate Control Scheme for Video Streaming over Wireless Networks," Proc. of the 3rd Inter. Conf. on Image and Graphs (ICIG '04), IEEE Computer Society, 2004.

Conclusion:

NO This research has dealt with the effects of packet size on the video quality in terms of temporal scalability (frame per sec) over wireless Internet. We have applied a packet loss model based TCP-Friendly Rate Control (TFRC) for low bit rate MPEG-4 video streaming over an AWGN wireless channel. The model has estimated the effective playable frame rate (PFR) and its corresponding PFR% ratio. As a result, the results highlight that the proposed model based-TFRC introduces a good tolerance in video quality over wireless link as compared to the non-TFRC mode. Furthermore, the simulation results provide a direct guideline to track the effects of small packet sizes over radio link layer based TFRC. However, future work can involve channel coding and adaptive modulation in order to improve the low value range of SNR over the wireless channel; and eventually to provide more perceptive video quality at client in case of large values of packet size when a bad channel state is considered.

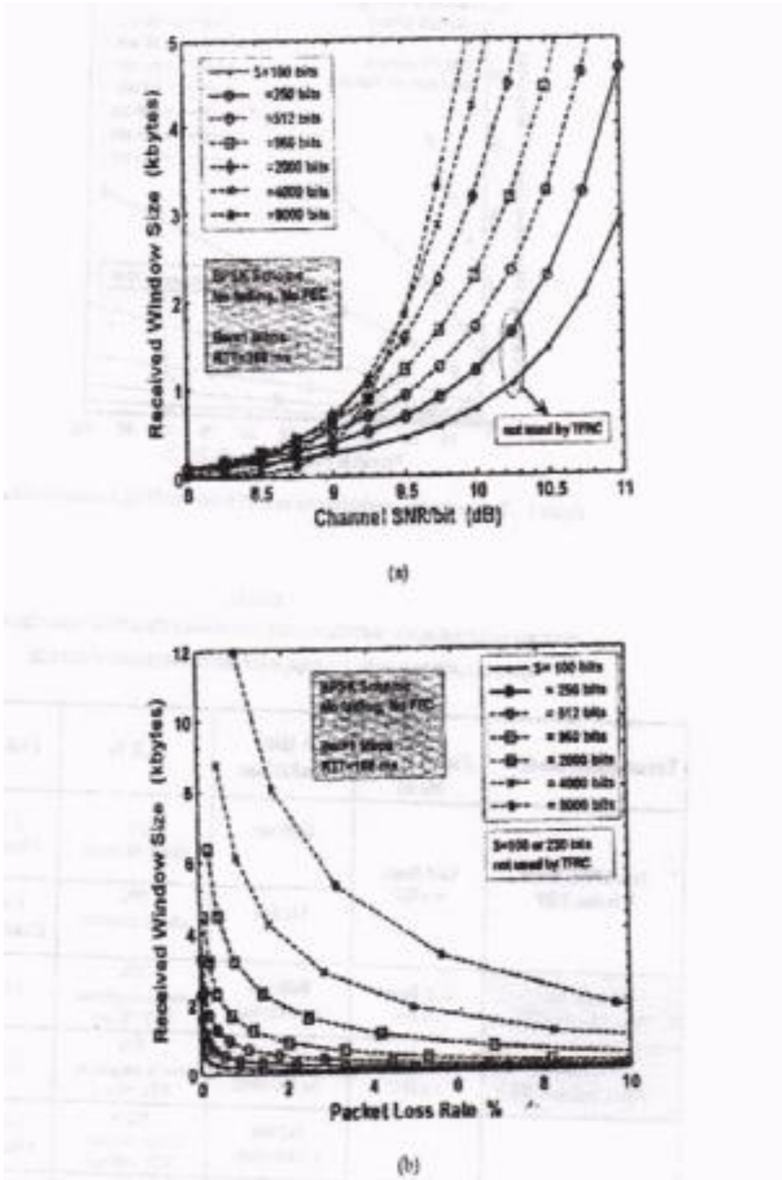

Figure 5: TFRC performance of only single video stream at the client for various values of packet size

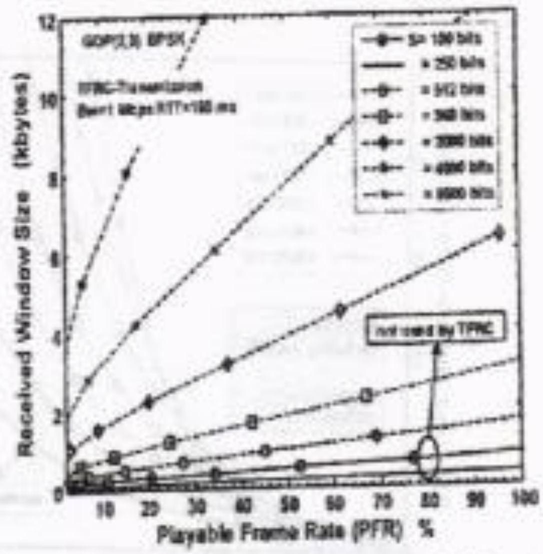

Figure 7 Temporal video scalability in terms of PFR ratio for TFRC transmission Mode

TABLE I

PERFORMANCE COMPARISON AMONGST PACKET LOSS MODELS FOR MPEG-4 VIDEO STREAM WITH PARAMETERS GoP(2,3), F_s =25 [fps] AND VARIOUS VALUES OF PACKET SIZE

Transmission-Mode	Packet –Loss Model	S (bits) Packet Size	PLR %	PFR (fps), PFR%
Non-TFRC Mode Wireless UDP	GoP-Based w/o FEC	8000 bits	5% (due to bit errors)	6.35 [fps], 25.3% Channel SNR (9.8 dB)
		512 bits	5% (due to bit errors)	6.32 [fps], 25.2% Channel SNR (8.43 dB)
TFRC-Mode Wired Internet [7]	GoP-Based w/o FEC	8000 bits (or 1 Kbytes)	5% (due to congestion, RTT~50 ms)	1.58 [fps], 6.32%
TFRC-Mode Wired Internet [8]	Frame-Based w/o FEC	8000 bits (or 1 Kbytes)	5% (due to congestion, RTT~50 ms)	1.25 [fps], 5%
Proposed TFRC-Mode Wireless Internet	GoP-Based w/o FEC	512 bits (or 64 bytes)	0.2% (due to bit errors, RTT~158 ms)	Full-motion 100% Channel SNR (10 dB)
		960 bits (or 120 bytes)	3.3% (due to bit errors, RTT~168 ms)	1.26 [fps], 5% Channel SNR (9.0 dB)
		8000 bits (or 1 Kbytes)	3% (due to bit errors, RTT~168 ms)	1.4 [fps], 5.6% Channel SNR (10 dB)

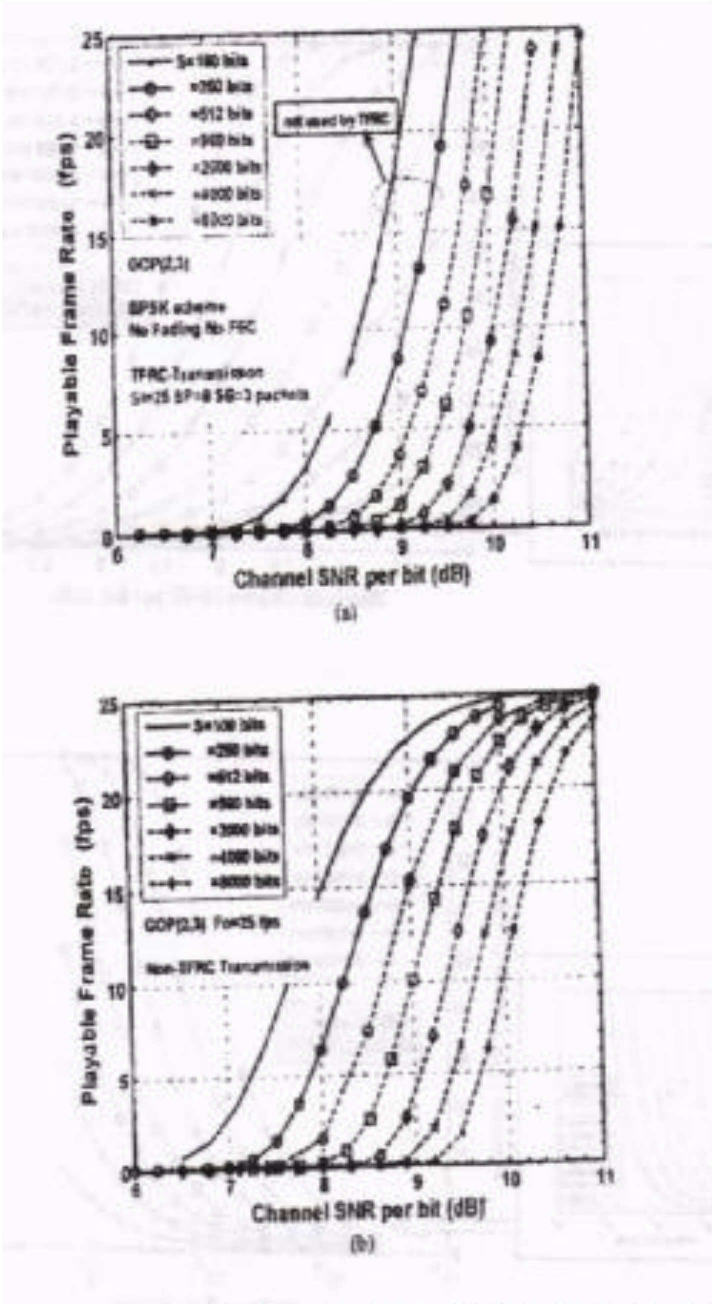

Figure 6: Temporal video scalability versus channel SNR/bit (a) based-TFRC Mode (b) based-Non TFRC Mode

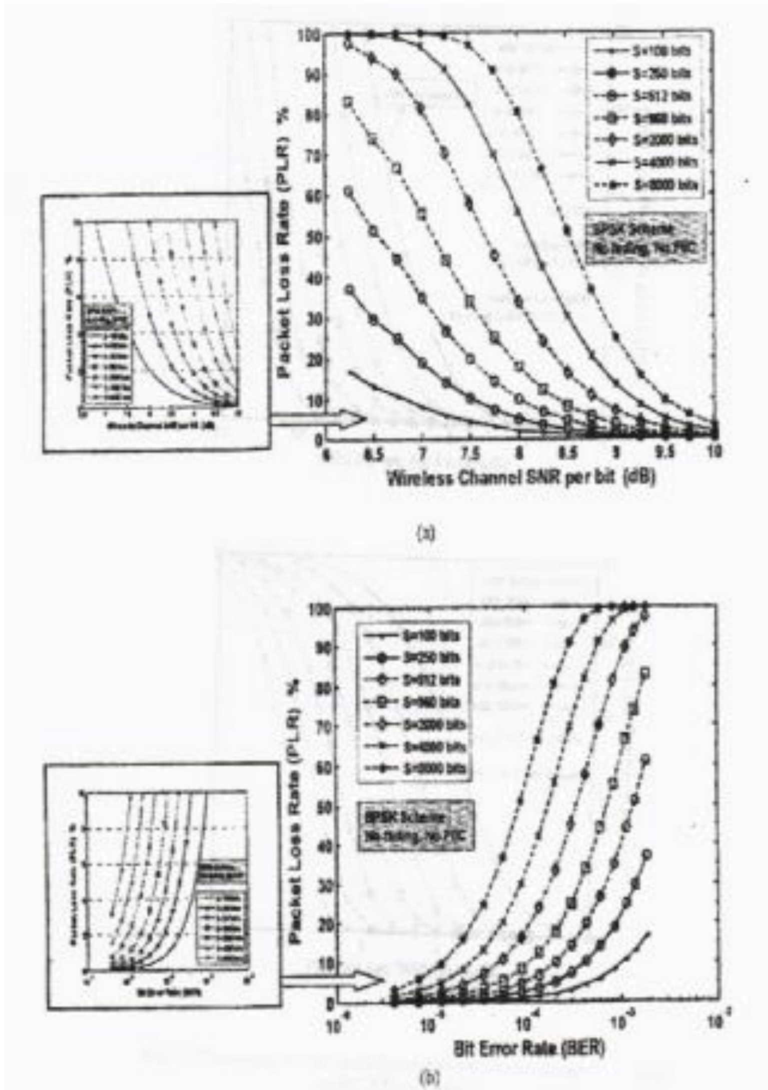

Figure 4 The performance of wireless channel versus packet loss rate for various values of packet size

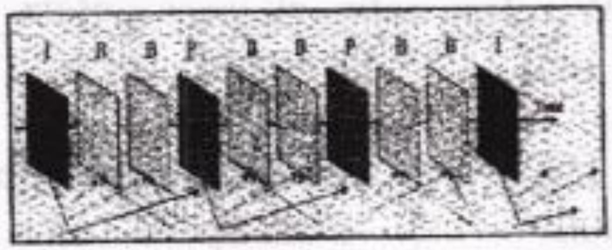

Figure 1 A structure of a GoP and inter-frame dependency relationship

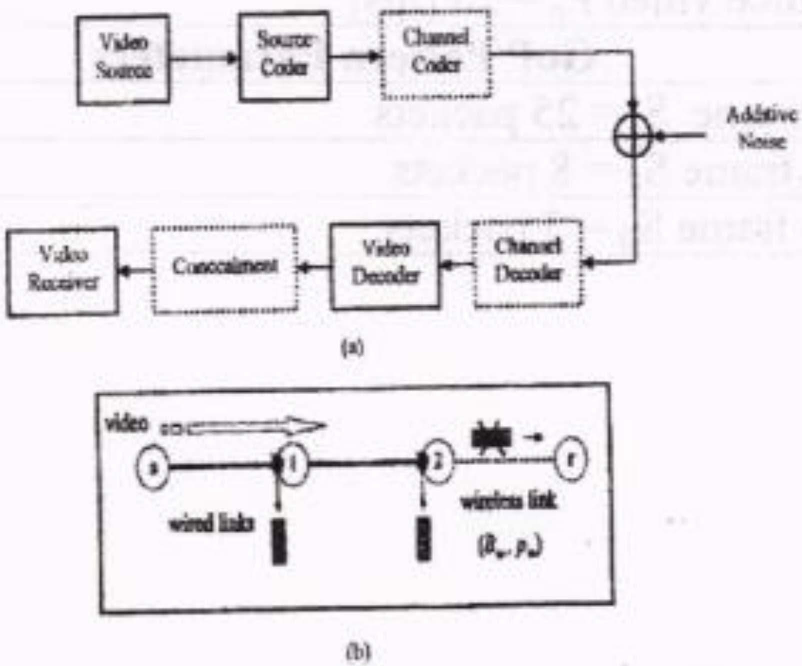

Figure 2 Video transmission system (a) A wireless channel model (b) A typical wired/wireless video streaming model

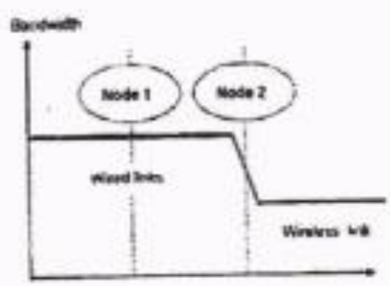

Figure 3 Bandwidth condition for wired-to-wireless video streaming model

Table I

Parameters setting used in simulation at node (2)

<table>
<tr><td colspan="2">Wireless network parameters</td></tr>
<tr><td>T_{RTT} = 168 [ms]
$T_{RTO} = 4t_{RTT}$
S = 64,120,250, 500, 1000 bytes</td><td>Bw= 1 Mbps (1 × RTT CDMA)
BPSK (uplink / downlink)
Data Speed 144 Kbps</td></tr>
<tr><td colspan="2">Channel SNR / bit y 6…….. 11 [dB]
Source video F_O = 25 [fps]</td></tr>
<tr><td colspan="2">G_oP Pattern Parameters</td></tr>
<tr><td colspan="2">I - frame S_1 = 25 packets</td></tr>
<tr><td colspan="2">P - frame Sp = 8 packets</td></tr>
<tr><td colspan="2">B - frame SB = 3 packets</td></tr>
</table>

transmission. Consequently, the play-out frame rate can achieve high values quickly when channel SNR becomes high enough using BPSK scheme over wireless link.

2. Since TFRC mode is originally UDP controlled by TCP equation model at the sender, hence, packet size can be typically chosen not less than 512 bits (i.e. 64 byte); to provide the fairness with TCP flows [5] over channel (by meaning the header of TCP packet is at least 20 bytes).

3. Since the header of UDP packet is typically 8 bytes in case of non-TFRC, then small values of S can be less than 512 bits. For example, setting S to be 100 bits will provide a highest video quality (full motion) at about 11 dB as shown in Figure 6 (b).

4. A full motion can be achieved at 0.2% packet loss rate (channel SNR 10 dB) when S is chosen to be 512 bits in case of TFRC-mode; and roughly at 11 dB in case of non-TFRC (See Figures 5 & 6 & Table 2).

5. TFRC-mode takes into account the network constraints such as network throughput; then the maximum channel SNR required for full motion will not be less than 10 dB and 11 dB when packet size changes from 512 bits to 8000 bits, respectively.

6. The non-TFRC is based on the prior probabilities of frame dropping in terms of PSNR of perceived video stream before video transmission starts. Hence, the maximum channel SNR required for full motion will roughly be much greater than 10 dB when S changes to large values.

7. Although the large values of S trends to increase the received window size at client, the successful PFR % remains lower than that of small values of S in case of TFRC-mode (See Figure 7). a to zawols shivorg

from large values (e.g. 8000 bits) to a small value (e.g. 512 bits) in a typical chosen GoP (2,3) pattern, this will provide a rapid full motion (high video quality) for video at client. In effect, to guarantee the video quality (5 to 25 fps), the corresponding channel SNR can be between 9.2 to 10 dB in case of 512 bits, and between 10.3 to 11 dB in case of 8000 bits.

Figure 6 (b) displays also a resultant PFR which is

based on non-TFRC transmission mode. The PFR can introduce a good tolerance for packet loss rate in terms of channel SNR, When packet size is chosen to be 100 bits, then the corresponding channel SNR can be between 7.25 dB to 10 dB for 5 to 25 fps, respectively. However, this range can be greater than 10 dB in case of packet size of 8000 bits.

Furthermore, Figure 7 explains the required window size at client as a function of PFR% in case of TFRC-mode for various values of packet size. It is noticed that a small packet size, for instance 512 bits; will need small window size, however, the temporal scalability (in PFR %) can rapidly achieve high values of full motion when the channel SNR becomes around 10 dB. In contrary, as S increases highly to be 8000 bits then there is a significant degradation in PFR % ratio although the received window size is being high at client.

As a result, in both modes, the study introduced a new guideline to investigate the effect of packet size of MPEG-4 video on temporal scaling (video quality) over wireless channel as compared with other recent studies for wired Internet [7,8] (See Table 2). The results can be summarized as follows:

1. A proper choice for packet size (i.e., by setting small values at node 2 in radio link transport layer) will provide a lowest or a minimum packet drop rate during

5.2 Results Analysis

Simulation results have been conducted for a typical wireless network model. Table 1 introduces a typical parameters setting for wireless network and GoP pattern parameters [5,7]. A channel capacity is assumed at the limited bandwidth (Bw), which represents a maximum throughput for wireless CDMA link.

In Figure 4, the performance of the wireless channel in terms of packet loss rate (PLR) is clearly evaluated through the change in the channel SNR per bit and its corresponding bit error rate for various values of packet size. It is noticed that as packet size increases, then the overall resultant PLR highly raises. For example, pe = 1 x 104 introduces PLR of 5% for 512 bits packet size as the results in [13]. Therefore, Figure 5 draws the effect of packet size on TFRC performance in terms of TCP-Friendly throughput and its related window size (in Kbytes), which is received at the client terminal. The window size is defined by the network throughput-Delay product; and the tRTT can be considered as the minimum effective delay on network. Hence, it is found that although the packet size S becomes large, the expected PLR increases due to the bit errors occurrence and the expected window size will remain also large. As a result, such large values of S cause a poor perception for video quality at the receiver in case of low values of channel SNR ratios.

Figure 6 (a) shows a significant improvement in the resultant PFR when S is chosen within small values (i.e., when S is being 512 bits) for TFRC mode. It is found that small packet size allows low resultant PLR and it also provides an opportunity for wireless channel to operate at a lower range of channel SNR as compared to the SNR range of large values of S.

Specifically, at node 2 when packet size S changes

5. Simulation Results

5.1 Methodology

To find the resultant TFRC-playable frame rate (PFR):

1. The video source initially starts with the packet size S at 1 Kbytes in wired links.

2. As soon as the video flow has arrived node 2, a bottleneck bandwidth problem will allow the video system to reduce its packet size before the wireless transmission starts.

3. At node 2, an appropriate low value of S should be chosen (such as 512 bits, 640, 960, etc.) depending on the wireless channel state in order to ensure a desired quality of video streaming at client,4. Thus the video system obtains a channel state in terms of SNR per bit (γ / 2) and assesses the bit-error rate pe by (4) on the wireless link for non-coded BPSK modulation scheme.

5. Then, it estimates the corresponding packet loss rate (PLR) using (2). Hence, the packet loss rate over a wireless link can be defined as pw = Pe

6. Estimate TFRC rate for MPEG video, which must satisfy the condition of the obtained pw equals Pe, by (3). Then, we can determine the video quality in terms of the temporal scalability, i.e., playable frame rate (R) by (8).

7. Non-TFRC transmission mode over wireless link can also be evaluated using (9) if the prior probabilities of I-, P-, and B-frames are per-defined at node 2 before UDP transmission starts. Hence, the predicted frame rate over network is estimated by using (10) if the original video frame rate is 25 [fps].

the frame is considered useless, i.e. dropped.

The strategy in this model is to assume that the network will depend on the pre-defined probabilities at video source of node 2 in Figure 2 (b). Inconsequence, the model can choose a GoP pattern before UDP transmission starts in order to obtain the reasonable expected playable frame rate that are compatible with the fali video motion at the client terminal end.

4. Performance Evaluation

To evaluate the received window size at the client in (Kbytes), a TFRC bandwidth-delay product can be considered. Here, a propagation delay is based on only fixed round-trip-time. As this product increases due to the network throughput then the expected video quality may eventually enhance at the client in terms of playable frame rate. In TFRC mode, this product is affected by the packet length over wireless channel.

Furthermore, one metric used to evaluate the video quality performance is the successful playable frame rate in (8) or (10), which can be also expressed as a ratio (in %) as follows.

PFR (in%) = (11) R Fo × 10

(12)

Where Fo is the source frame rate in frames / sec. Equation (12) provides the percentage of the frame rate in a GoP that can be decoded correctly at the receiver [8]. In practice, the source frame rate typically varies between 15 and 30 [fps] depending on the applications. In the simulation results in Section 5, we assumed 25 [fps] is as a reference frame rate (Table 1).

Pem (1) can be defined as in (2). Thus for non-coded BPSK mode, the coding rate is not taken into account, and the factor A should be known in terms of channel SNR in order to evaluate the effective network throughput. On the other hand, Equation (9) can be reformatted in terms of frame-dropping mechanism in order to explain the variability in the original video sending rate due to packet loss over wireless link ignoring the effect of any congestion at node 2. Hence, the frames are assumed to be dropped, or lost, by corruption of packets due to bit errors on wireless link. If the frame quality in terms of Peak Signal-to-Noise Ratio (PSNR) falls below a predetermined threshold PSNRthreshold then the frame is considered lost at the receiver (i.e., mobile side).

$$R_{non}\text{-}_{TFRC}=F_o\ (1-\acute{\O}_R), \qquad (10)$$

Where & stands as an effective "frame drop rate", i.e., the fraction of frames dropped, and F, [fps] is the frame rate of the original video stream. If quality scaling is applied, a constant F. can be employed to support non-TFRC transmission mode. At node 2, the frame drop rate can be predicted and formulated as a sum of conditional probabilities as [6],

$$\acute{\O}_R = \sum P(f).P(F\backslash f_1) \qquad (11)$$

i Where i runs over the three frame types (I, P, and B), F represents the event that a frame is "useless" because the quality falls below quality threshold PSNRthreshold and fi, is the event that the type of the frame is i. The a priori probability P(f) can be determined directly from the structure of a stream [6]. The conditional probabilities for each frame type of size St, Sp, and SB can be derived under the assumption that if one or more packets within a frame are lost or one or more packets are lost in a reference frame,

3.3.1. GoP Rate 5091

Under the TCP-Friendly constraint of (1), the GoP rate G (in GoP per second) is computed as,

$$G= \frac{T_b IS}{S_I + N_P S_P + N_B S_B} \qquad (6)$$

Where G corresponds to the number of GoPs per second, S1,Sp, and Sß are the frames' sizes of the I, P, and B frames in GoP pattern (in packets). Then the GoP size can be expressed as,

$$S_{GOP}\ \text{size} = 1 + N_p + N_B \qquad (7)$$

3.3.2. Total Playable Rate

The total playable frame rate can be evaluated as [7],

$$R_{TFRC} = \sum R_1 = R + R_p + R_B \qquad (8)$$

3.4 Packet-Loss Model Based non-TFRC

When UDP protocol is considered to stream MPEG-4 video over wireless channel, an extra control algorithm is required to control the congestion over wired-to-wireless links. In the model of Figure 2 (b), we assume the following scenario. The packet size starts with 1Kbytes at the video source, and this packet size should be changed at the node 2 to avoid the congestion (by meaning P. = 0) at the base station (BS) of the wireless network. Furthermore, a proper packet size will introduce a significant effect on the network performance. Hence, the effective physical layer throughput can be expressed as [11],

$$T_{phy,m} = A\,(1 - Pc,m\,(I)\,) \qquad (9)$$

The factor A represents the maximum achievable data rate in (Kbps) for mode m. The probability of packet error

[5,9]. The backward route from receiver r to server s is assumed to be congestion-free but not error-free due to bit errors, hi this scenario, the video sending rate is smaller than the bottleneck bandwidth (i.e. Bw) and should not cause any network instability, i.e., congestion collapse. Additionally, the optimal control should result in the highest possible throughput and the lowest packet loss rate. To derive the target sending rate which satisfies them by using (1), packet loss rate p is now defined by two independent loss rates pw and pc as, p = pw+(1-Pw) Pc. Since Pw gives the lower-bound for p if pe = 0, the upper-bound of the network throughput becomes,

$$T \leq \frac{s}{t_{RTTmin}\sqrt{\frac{2p_w}{3}} + t_{RTO}\sqrt{\frac{27p_w}{8}}\,p_w\left(1+32p_w^2\right)} = \quad (5)$$

(8) Hence, for an under-utilized channel, $T_{\flat} < Bw$ holds when only one TFRC connection exists.

Here, the reported pw at the receiver depends on the source packet length (size) before TFRC transmission starts. To estimate the number of playable frames at a receiver, random and stationary packet losses are considered over wireless network. Thus the analytical model designed over wired Internet for MPEG-2 video stream, which is proposed by Wu et a 1. [7], is applied in this paper for GoP pattern of MPEG-4 video stream over wireless link. This model basically employs TFRC protocol to control the sending rate in accordance with loss of packets caused by packet corruptions for bit errors over a wireless channel. Subsequently, a GoP rate can be analytically expressed using TFRC protocol and the frame dependency relationship of I, P, and B frames. Hence, the resultant playable frame rate R can be computed as follows.

here the BER performance of non-coded BPSK scheme over AWGN channel has ignored the channel coding, and the probability of bit error is given by [I],

$$P_b = Q\left(\sqrt{\gamma}\right) = Q\left(\sqrt{\frac{2E}{N_c}}\right) \tag{4}$$

Eb stands for the bit energy, No is the noise power, and y = 2E / No represents the total channel SNR of a non-coded BPSK scheme assuming there is no channel encoding for this modulation scheme. The Gaussian cumulative distribution function is being Q (.) and can be approximated in a close formula for a limited range of y.

3.3 Packet loss Model Based -TFRC

Because TFRC was initially designed for wired networks, when applied to wireless environment, the original TFRC receiver cannot distinguish the congestion packet loss from the wireless packet loss. Thus the loss event rate p calculated by the receiver consists of wireless bit errors part and congestion part. However, TFRC sender actually only needs the congestive loss event rate, so it may result in bandwidth underestimation if the original loss event rate is directly used. Hence, our solution is to follow this scenario.

When there is no cross-traffic at either node 1 or node 2, this scenario can be applied as follows. The wireless link is assumed to be bottleneck of the network by meaning no congestion at node 1 as shown in Figure 3. Packet loss is assumed only due to wireless channel bit errors and the buffer at node 2 does not overflow, as p = 0. In consequence TRTT = TRTT min, i.e., the minimum RTT, if T < Bw. Here, Bw is assumed limited constant bandwidth and pw is to be random and stationary packet loss due to bit errors loss rate Pw due to bit errors. Meanwhile, Po at node 1 and /or node 2 denotes the packet loss rate due to the congestion (buffer overflow). In this model, a source node cannot distinguish packet losses caused by bit errors on wireless links from those caused by buffer overflow [1,5]. Thus we consider only the bit error rate over wireless link is the considerable reason to generate this packet loss.

3.2 BER Performance

At physical layer, to obtain Pw, a frequent bit error of a wireless channel with AWGN is considered. BPSK scheme is applied and the fading effect is ignored. Within proposed model, we will refer to the term "mod m" to indicate to a specific choice of a modulation and coding scheme. The probability Pe.m (I) of error in a packet of length 1 bytes (also referred to as the physical layer packet loss rate, PLR), for a given mod m, as a function of the bit error probability P_1 be expressed by [11], Pe,m (I) ≤ 1-(1- Pb,m)81 (2) Where S = 81 denotes a packet size (in bits), and the inequality in (2) represents the fact that one can recover from bit errors in a packet, due to the coding scheme used at the packet level (intra-protection). Also, the packet error probability in (2) can be denoted as packet loss rate without any error-correction procedure when the inequality is replaced by equality [2]. In error-correction code with maximum error capacity t is used, $P_{ec,m}$ can be expressed as,

$P_{ec,m}$ = 1 – Στo (81) Pom (1 – Pb,m) ^{81}

i (3)

A typical performance can be improved when the effectiveness of FEC coding is taken into account. In this paper, we simulated the results ignoring the procedure of (3) in order to only specify the effective operating ranges of channel SNR ratios at various packet lengths. Therefore, TRTT is the round-trip time [sec], TRTO is the TCP retransmit time out value [sec], S is the packet size [byte], and p is the loss event rate reported by the receiver. By regarding T as the available bandwidth for video streaming and adjusting the video traffic, the high-quality video play-out at a receiver can be expected.

3. Proposed Approach

In this paper, two packet loss models for MPEG-4 video streaming over wireless Internet are investigated on AWGN wireless channel; one model is under TFRC transmission mode (i.e. variable frame rate depends on TFRC throughput over network), and the other model is assuming a fixed or constant playable frame rate during non-TFRC transmission mode, i.e. using UDP transport protocol.

3.1 Wireless Channel Model

A realistic wireless video transmission system can be represented by the model shown in Figure 2 (a). 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 [1].

In this paper, the concealment stage and the channel coding are not considered in our proposed approach. Thus we assume Figure 2 (a) is a part of a typical model of video streaming over wired and wireless links in Figure 2 (b); whereby a video server S in a wired network sends a video stream to a receiver r behind a wireless link. The wireless link is characterized by available bandwidth Bw and packet proposed approach, and followed by Section 4 for performance evaluation. Simulation results are explained in Section 5, and finally conclusions are summarized in Section 6.snimk

2. Background:

2.1 MPEG Video

Moving Picture Expert Group (MPEG) is a popular standard for video compression. Figure 1 illustrates a typical Group of Pictures (GoPs) structure of an MPEG stream and its frames' dependencies. A GoP consists of three types of frames: I-, P- and B-frames. An I- frame (mtra coded) located at the head of a GoP is coded as a still image and serves as a reference for P and B frames. P-frames (Predictive coded) depend on the preceding I or P-frame in compression. Finally, B-frames (Bi-directionally predictive coded) depend on the surrounding reference frames, that are the closest two I and P or P and P frames [7]. In general, a GoP pattern for MPEG-4 video can be identified in similar manner of MPEG-2 video, as G(NP, NBP) and NB = (1+ Np) x NBP, where NB corresponds to the total number of B-frames, Np corresponds to a number of P-frames in a GoP, and NBP corresponds to the number of B-frames between I and P frames, such as GoP(2,2) “IBBPBBPBB" where Np = 2 and NBP = 2.

2.2 TFRC Equation Model

In the original TFRC [5], the sender uses TCP throughput model to estimate the current available network bandwidth, i.e.,

$$T = \frac{S}{t_{RTT}\sqrt{\frac{2p}{3}} + t_{RTO}\sqrt{\frac{27p}{8}}\, p\,(1+32p^2)} \qquad (1)$$

Where T is the target sending rate of a TFRC session, estimation of the amount of distortion. Yaun et al. [8] proposed a FEC scheme in the GOP-level with cost of high complexity of the packet generation. Furthermore, Feamster et al. [6] proposed error-model for MPEG-4 based on constant frame rate using UDP and I-frame retransmission. Finally, AL-Suhail et al. [9] have applied a Wu's model [7] over wired/wireless channel and improved the QoS using FEC in the packet-level. Additionally, other studies [1,12,13] combined adaptive modulation and joint source channel coding over fading wireless channels and verified significant performance advantages in the worst channel conditions. Zahi [1] proposed optimal cross-layer resource allocation over wired/wireless link for real-time video transmission without taking into account TFRC protocol. In contrary, Khan et al. [12] proposed CLD for wireless video streaming based on Peak Signal-to-Noise ratio (PSNR) versus the communication cost associated with the transmission of rate-distortion profile from the video server. Meanwhile, Tong et. al. [13] presented TFRC-LERD scheme using loss event rate discounting to improve TFRC over wireless networks.

In this paper, we assume that the physical layer of wireless channel can estimate the performance of symbol or bit error rate (BER) versus Signal-to-Noise Ratio (SNR) when the channel is corrupted by an Additive White Gaussian Noise (AWGN). Thereby a Binary-Phase-Shift-Keying (BPSK) modulation is chosen to provide a robust transmission against high bit errors. By using packet-loss model, a video playable frame rate can eventually be predicted in the application layer when an appropriate small packet size is chosen well for TFRC transmission mode as well as in case of non-TFRC.

The rest of the paper is organized as follows. Section 2 describes a brief background. Section 3 presents the alternatives to UDP over wired/wireless networks. There are three advantages to rate control using TFRC: first, avoid congestion collapse (providing network stability); second, it is fair to TCP flows and third, TFRC's rate fluctuation is lower than TCP, making it more appropriate for streaming applications. However, the key assumption behind TCP and TFRC is that the packet loss is a sign of congestion, whereby neither TFRC nor TCP can distinguish between packet loss due to buffer overflow and that due to physical channel errors, resulting in underutilization of wireless bandwidth. Hence video streaming rate control and congestion control over wireless are still open issues.

In traditional communication system, channel variations are dealt with in a worst-case manner. For wireless systems this implies the use of a simple modulation scheme, and a complex error-correcting code. When the coding fails to compensate for temporary bad conditions, higher layers in the protocol will ensure that the information is correctly and completely transmitted, by requiring a retransmission of the erroneous data. Therefore, to avoid this problem adaptive modulation schemes are used to adapt the demands on the channel as it varies. By changing the modulation format as the channel SNR (or SINR) varies, this will accomplish less retransmission.

Therefore, to provide an acceptable video quality; i.e. high-Playable Frame Rate (PFR), at high loss rates of wireless links, several approaches have been either separately or jointly pursued. They include adaptive rate control, passive error recovery (re-transmission), Forward Error Correction (FEC), and adaptive modulation [6-13]. For example, Wu et al. [7] extended the packet loss model for MPEG-2 video streaming and derived analytically the playable frame rate (PFR) for a given packet loss probability over wired Internet. They employed VQM method to regulate the level of quantization via the

1. Introduction:

Recently, the demand on multimedia applications is widely increased over wired and/or wireless Internet, including such as real-time video streaming, video conference, and video on demand. However, wired /wireless Internet does not provide the necessary Quality of Service (QoS) guarantees that are needed to support high-quality video transmission. Many major challenges of video traffic on the wired and wireless Internet links [1-2] are encountered. Some of these challenges deal with high packet loss rate due to the congestion of buffer overflow over wired networks; and others are faced by the characteristic of wireless links, which are mostly suffering from low bandwidth and high error rates due to the noise, interference, fading and shadowing. The bit stream video over a noisy channel introduces symbol or bit errors causing packets corruption, which leads to a significant degradation in the quality of reconstructed video sequence. Thus robust transmission of real-time video over wireless channels is becoming critical problem to achieve good perceptual quality at the client terminal end.

Unlike typical Internet traffic, streaming video is sensitive to delay and jitter, but can tolerate some data loss. In fact, video transmission can yield better video play-out when the underlying protocol provides smooth data rate than a bursty data rate. For this purpose, video streaming applications often use UDP (non-TFRC) or TCP-Friendly Rate Control (TFRC) as a transport protocol rather than TCP. Unfortunately, UDP does not reduce its data rate when an Internet router drops packets to indicate congestion. It means that there is no congestion control within UDP and no response to relieve the saturation of a bottleneck due to congestion. Thus recent researches have proposed rate-based TCP-Friendly protocols for steaming media [3-5] as

The Effect of Band's Size on the Temporary Qualities for Transferring Video Pictures Wirelessly

Alharith. A. Alkafije

College of Science

University of Babylon

Abstract: In this research, the sample of band 's reducing had been applied with analyzing for some pictures of (Gops) sample (MPEG-4) through internet. suppose the sample AN AWGN wireless channel caused small followed errors corresponding with the bands, therefore; the effect of the band's size is the first to be tested with qualities of temporary video in the case (TFRC) controlling range(TCP) then Comparing the quality of video in the case of transferring (TFRC).

The new supposed results had appeared that it is possible to assess the performance of video qualities, it is expected through the percentage of the picture which is liable to treatment (picture ,second). At the different level for single channel to the percentage of annoyance (in dBs) when the size of band would be chosen before the transferring. Besides to the similarity the quality of releasing video on of wire net.