

## **A Cross-Layer Model for Video Multicast Based TCP- Adaptive FEC over Heterogeneous Networks**

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## **Abstract**

In this paper, we present a simple cross-layer model that leads to the optimal throughput of multiple users for multicasting MPEG-4 video over a heterogeneous network. For heterogeneous wired-to-wireless network, at the last wireless hop there are bit errors associated with the link-layer packets that are arising in the wireless channel, in addition of overflow packet dropping over wired links. We employ a heuristic TCP function to optimize the cross-layer model of data link and physical (radio-link) layer. An adaptive Forward-Error-Correction (FEC) scheme is applied at the byte-level as well as at the packet-level. The corresponding optimal video quality can be evaluated at each client end. The results show that a server can significantly adapt to the bandwidth and FEC codes to maximize the video quality of service (QoS) in terms of temporal scaling when a maximum network throughput for each client is reached.

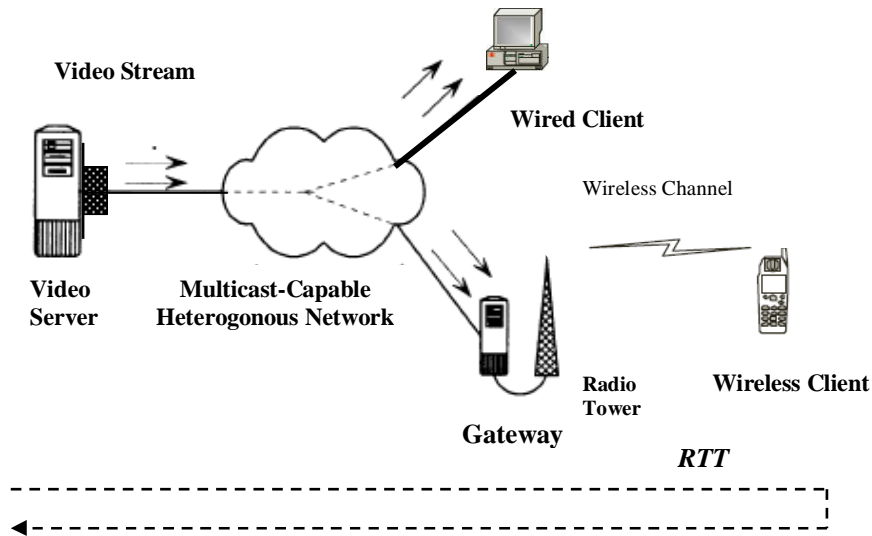
# **A Cross-layer Model for Video Multicast Based TCP-Adaptive FEC over Heterogeneous Networks**

## **INTRODUCTION**

Video multicasting has become increasingly deployed in many multimedia applications (e.g., Telemedicine systems, Video on Demand, etc.), which involve point to multi-point communication, i.e. a video sequence stored or generated (captured live) at a server is simultaneously delivered to a group of receivers distributed in a network (Bajie, 2006; Martini et al, 2007; Liu et al, 2007). In such a network, packet loss is inevitable. In order to provide good video quality, it is important to recover most of the loss so that the resultant end-to-end recover error rate after correction, i.e., the residual loss rate, is kept below a certain value (Chan et al, 2006).

Fortunately, video multicast is an efficient way to deliver one video simultaneously to many users over homogeneous and/or heterogeneous wired-to-wireless networks, such as in wireless IP applications where a mobile terminal communicates with an IP server through a wired IP network in tandem with a wireless network as in Figure 1. Compared to unicast, it improves bandwidth efficiency by sharing video packets delivered through network. However, it suffers some particular problems arising from the use of wireless network applications. For example, a multicasting wireless network is often characterized by having a physical channel that is highly error-prone and time-varying. In addition, users in such a network can often have diverse channel conditions (Liu et al, 2007; Lo et al, 2005; Lee et al, 2002; Pei et al, 2004).

On the other hand, many popular multimedia networks cannot provide a guaranteed of quality of service (QoS) for video traffic. MPEG-4, however, is still a video compression standard adopted by most mobile and wireless networks because of its good quality at bits rates of these networks (Lo et al, 2005). To this end, it is essential to rely on QoS metrics of a connection (flow or session) in terms of data throughput, packet error/loss rate, and delay performance especially over heterogeneous networks. In practice, for QoS guarantees in high-rate multimedia applications, many major challenges of video traffic are faced on heterogeneous wired and wireless Internet links (Lee et al, 2002; Pei et al, 2004; Liu et al, 2004; Zhang et al, 2006; Chiasserini and Meo, 2002). 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 is mostly suffering from low



**Figure 1.** Video multicast system over heterogeneous wired-to-wireless Network

bandwidth and high bit error rates due to the noise, interference, unpredictable user mobility (Doppler effects) and multi-path fading. Specifically, the “bottleneck” common to both (military or civilian) networks is the wireless link, not only because wireless resources (bandwidth and power) are more scarce and expensive than their wired counterparts, but also the overall system performance degrades markedly due to time- and frequency-dispersive fading 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 (Chen et al, 2006). These transmission rates decreases are unnecessary and lead to resource inefficiency. Unlike wired networks, even if large bandwidth is allocated to a certain connection, the loss and delay requirements may not be satisfied when the wireless channel experiences deep fades or high noise. Hence, a powerful forward error correction (FEC) coding or automatic-repeat request (ARQ) protocols can reduce the loss rate, at the expense of increased bandwidth and delay (Pei et al, 2004; Liu et al, 2007). However, allocating a fixed amount of bandwidth to each user may not be as efficient, the queues may be empty from time to time due to the dynamic nature of the traffic. As a result, the difficulty in simultaneously guaranteeing QoS and utilizing resources efficiently can be testified due to these given considerations (Lee et al, 2002). Therefore, there are generally two types of solutions to solve this problem over wired-to-wireless networks: (i) modifying the TCP model, e.g. (Akan et al, 2004), and (ii) improving the link reliability observed by TCP (Liu et al, 2004; Zhang et al, 2006; Chiasserini and Meo, 2002; Zorzi et al, 2002; Barman et al, 2004; Chen and Zakhor, 2006; Xylomenos et al, 2007). The later solution mainly includes a cross-layer design of hybrid schemes to provide a required quality link reliability over wireless links. There exist many recent cross-

layer approaches in this direction involving, for example: adaptive rate control (Zhang et al, 2000; Chen and Zakhor, 2006), adaptive selective Repeat (ASR) protocols (re-transmission) (Xylomenos and Makidis, 2007), finite-length queuing at the data link layer coupled with adaptive modulation and coding (AMC) at the physical layer (Liu et al, 2004), and finally adaptive Forward-Error-Correction (FEC) at packet level (*e.g.*, RS code and/or channel bit and byte-levels (*e.g.*, BCH, RCPC and RS) such as (Lee et al, 2002; Pei et al, 2004; Liu et al, 2004; Zorzi et al, 2002; and Barman et al, 2004).

On the other hand, there has been much work on using error control for *unicast* video delivery (Bajie, 2006; Zhang et al, 2006; Zhang et al, 2000) [2,9,20]. We study video *multicast* here based cross-layer design. Several error-recovery schemes have been studied for video multicast such as the so-called hybrid ARQ-FEC, and sending delayed version of parity packets (hybrid FEC-Replicated Delayed version) over different multicast groups for heterogeneous wired and wireless networks, *e.g.* (Liu et al, 2007; Chan et al, 2006; Lee et al, 2002; Rubenstein et al, 1998). For example, an evaluation of mixed media (wired and wireless networks) has been pursued for optimal system operation when packet-level and byte-level FECs are applied (Chan et al, 2006; Lee et al, 2002). Only bandwidth allocation at receivers for layered multicast has been examined using a receiver-driven multicast system with joint bandwidth and FEC allocation for each video layer in order to maximize the overall video quality. The study observed that the video quality in terms of peak signal-to-noise ratio (PSNR) is proportional to the video goodput (maximum throughput obtained *after* error correction) defined as the useful data bits per second received at a client.

In general, packet recovery and FEC are two commonly used mechanisms to improve the throughput of a wireless network. However, there are drawbacks when applying them directly in the multicast scenarios. For example, with retransmission, it is inefficient to multicast one packet that is only lost by one user again to all the other users. Therefore, Automatic Repeat request (ARQ) is clearly not suitable for real-time multicast applications due to the recovery delay. Meantime, FEC can maintain a constant throughput and bound delay. But, to guarantee certain QoS, FEC codes are often used for the worst channel conditions. That may unnecessarily penalize the throughput when the channel is in good state (Liu et al, 2007). Moreover, since the final performance (*e.g.* optimal achievable throughput at the transport level which consequently introduces the optimal play-out rate at client ends) depends on the concatenated adaptive FEC schemes at video server based on the feedback of radio link layer for wireless clients, the consistent TCP model is used to capture interactions and simplify the design complexity. Therefore, along this line, this paper moves from recent papers, *e.g.* (Liu et al, 2004; Zhang et al, 2006; Martini et al, 2007; AL-Suhail, 2008) and proposes significant

extensions in order to jointly consider the following components in the optimization task: bandwidth allocation, modulation format (i.e., BPSK over AWGN channel), error recovery strategy via adaptive FEC codes at byte and packet levels using RS code, and finally temporal video quality in terms of frame per second.

In this paper, we propose a framework of TCP-Adaptive FEC scheme to improve the link reliability via achieving a maximum TCP throughput for each client of MPEG-4 video multicast over a hybrid network. Our cross-layer design considerably depends on the end-to-end bandwidth between a client and the server; a client (receiver) may progressively improve the video quality by adapting Reed Solomon (RS) FEC codes at the packet-level as well as byte-level at the data-link (radio-link) layer. The cross-layer model can eventually estimate the wireless channel state at the physical layer for each client and return feedback to the server to adapt the desired FEC codes. For sake of simplicity, time-invariant wireless channel is assumed corrupted by high error-bits in terms of channel Signal-to-Noise Ratio (SNR). As a result, the model can predict the quality of MPEG-4 video by adapting a playable frame rate (temporal scaling) under various packet errors and channel errors conditions.

The main difference between this paper and other cross-layer model papers is that the cross-layer model proposed for MPEG-4 video multicasting here is carried out for the end-to-end residual packet error when a certain system threshold of a residual packet error is setting at video server over wired-to-wireless networks. Our model basically uses TCP-adaptive FEC scheme to achieve the optimal video quality at wireless client ends in terms of temporal scaling (play-out frame rate) in place of only optimal or maximum TCP throughput which is mainly in most other works representing a Peak Signal-to-Noise Ratio (PSNR) in dB for MPEG-4 video sequences. In other word, the schemes of (Lee et al, 2002; and Chan et al, 2006) use a QoS metric for MPEG-4 video sequence based on adjusting the degree of quantization during the video coding process at the video server and eventually on the received Peak Signal-to-Noise Ratio (PSNR) at the client ends. However, the advantages of our model are as follows: first, it is an end-to-end simple approach and does not require any modification to network infrastructure and protocols, except some computations and look-up table (memory) at the video server based on the feedback from the client ends. Second, as will be pointed out later, it has the potential to allocate the required optimal bandwidth based adaptive FEC codes at byte-level as well as packet level. Third, it can effectively introduce a good play-out frame rate at client end either by adjusting a relevant system threshold of residual packet error required at the data-link layer of video server and/or adjusting a frame type of Group of Picture (GoP) pattern at application layer of video server to achieve the optimal or maximum video quality at the client end.

The rest of paper is organized as follows. In the next section, we will present the video multicast system model followed by our proposed cross-layer model involving the wireless channel model, adaptive TCP throughput formula, optimal FEC codes, optimal TCP-adaptive FEC over Wireless, and the temporal video model for multicasting MPEG-4 video. We will then explain some numerical result. Finally, we will summarize conclusions and introduce future work for more robust video multicast over wireless links.

## VIDEO MULTICAST SYSTEM MODEL

### Video Traffic

Although MPEG-4 (Lo et al, 2005) provides efficient and scalable video coding with constant bit rate (CBR) and variable bit rate (VBR) traffic, we consider the base layer of MPEG-4 video as a reference layer for all clients of a heterogeneous network. Any enhancement layer, for sake of simplicity and avoiding the complexity of dynamic programming (Chan et al, 2006), is treated as the base layer in our analysis. All clients' applications employ TCP over IP. In general, MPEG-4 server of a heterogeneous network generates three types of frame in terms of temporal processing: Intra-coded (I), Predictive (P), and Bi-directional (B). I frames contain the bulk of the audio and video data and are larger than P frames, which in turn are larger than B frames. When a video sequence is compressed, a typical MPEG encoder uses a pre-defined group of pictures (GoP), such as *I-B-B-B-P-B-B-B-P-B-B-B*, as used in our video model. However, as our cross-layer model basically use application layer information of MPEG-4 video, flows can have different GoP patterns (Wu et al, 2005).

### Network Model

Figure 1 shows the video multicast system considered in this paper for both wired and wireless clients. In case of wireless access, the base station is connected to a gateway. In its simplest form, the gateway forwards whatever packets it receives to the wireless clients (i.e., last wireless hop) without any re-packetization or fragmentation. In contrast, there are some sophisticated gateways can do some re-packetization (e.g., by adding or removing error redundancy codes) before forwarding the packets from the wired infrastructure to the end clients. The data packets are said to be "transcoded" in the process (Lee et al, 2002; Pei et al, 2004). This kind of "trans-coding" gateway may be beneficial since the error characteristics are different in the wired and wireless networks. In fact, in wired networks (such as Internet),

packets are dropped mainly due to congestion at the routers, while in the wireless hop, packets are often lost due to random bit errors caused by noise, fading or multi-path effect.

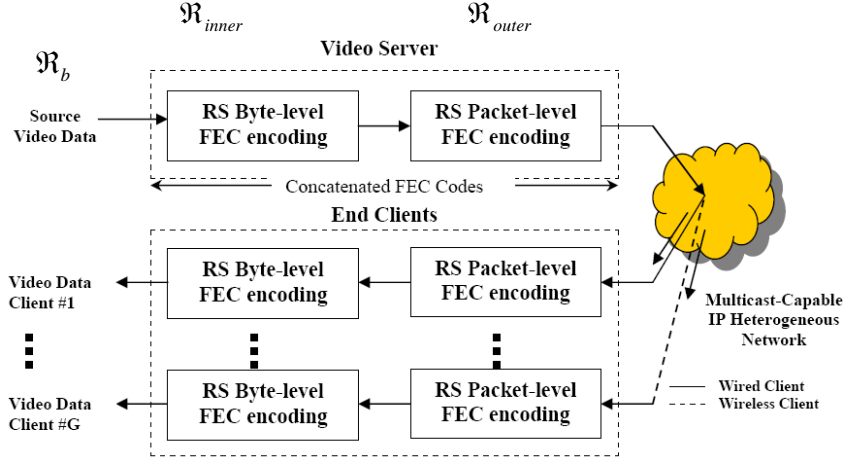
In this paper, we consider non-transcoding gateway as it performs only slightly better than the transcoding one in terms of end-to-end loss rate (Lee et al, 2002). Further, we intend to address an end-to-end solution for video transmission over a heterogeneous network such as the UMTS third-generation wireless system which provides the flexibility at the hardware radio-link layer to introduce optimal service-specific forward error coding as well as the necessary bit-rate required for high-quality video, up to 384 kbps (Pei et al, 2004).

## **Error Control**

In video multicasting, shown in Figure 2, due to the heterogeneous nature of channel conditions, i.e., bandwidth and error rate, among the clients and such conditions mainly varies over time, the video server has to continuously adapt the error-recover mechanisms and bit rate in order to optimize the overall temporal video quality in frame per second at clients. Specifically, to recover the packet loss, feedback recovery or forward-error correction code (FEC) may be used (Chan et al, 2006; Bajie, 2006; AL-Auhail, 2008). In general, feedback recovery does not work very well over long distance with real-time guarantee. When multicast groups grow large, simple reliable multicast protocols (Chan et al, 2006; Rubenstein et al, 1998) suffer from a condition known as *feedback implosion*: an overload of network resources due to the attempts of many receivers trying to send repair requests (referred to as NAK) for a single packet. A number of approaches exist to avoid this implosion effect such as randomized timers, local recovery whereby receivers can also send repair packets, and hierarchical recovery. However, while such approaches are effective providing reliability without implosion, they can result significant and unpredictable delays making them unsuitable for real-time applications which eventually need stringent real-time constraints. Moreover, many studies have devoted on hybrid schemes in combination of forward error correction (FEC) and automatic repeat request (ARQ) to reliably deliver data with an emphasis on reducing delay and meeting real-time constraints without using the solutions of randomized delays, local recover, or hierarchical recovery (e.g., Rubenstein et al, 1998; Marco et al, 2006; Liu et al, 2007). However, our framework in this paper is out scope of delay constraints of hybrid ARQ schemes for error recovery at the link layer.

In this paper, forward error correction (FEC), on the other hand, is more appropriate in this case (Chan et al, 2006; Liu et al, 2004). This scheme uses a code, i.e. an  $(n,k)$  linear code,





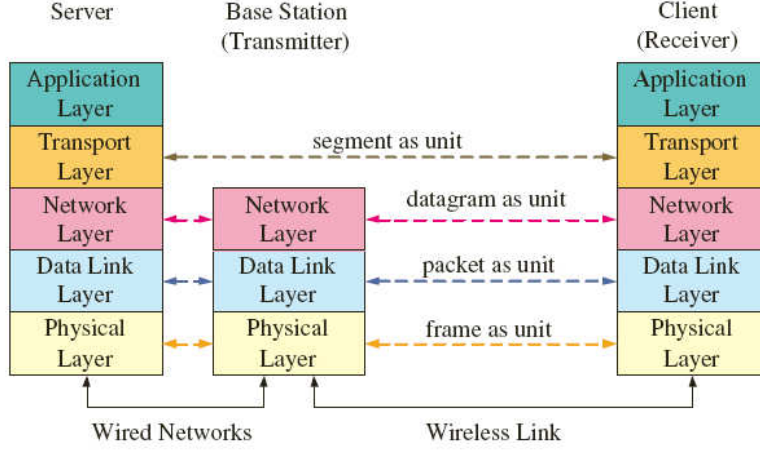
**Figure.2** Concatenated FEC codes for video multicast system over heterogeneous network.

which is designed for simultaneous error detection and error correction. It consists of arranging the data and redundancy symbols (bits) in such a way that even when not all the symbols (bits) are received, the original data may still be recovered. When the received block is detected with errors, the receiver first attempts to locate and correct the errors. If the presence of an error pattern is corrected that exceeds correction capabilities, the receiver rejects the received block and requests a retransmission. For our computations we adopted a Reed-Solomon (RS) code at byte and packet levels to provide a robust error cross-layer model for MPEG-4 multicasting over TCP transport layer. By adapting concatenated FECs according to the network conditions video quality at client can be efficiently maintained (Liu et al, 2002). Our model, in Figure 2, is effective in terms of FEC parity and allocated bandwidth required to each user (client) over multicast-capable IP heterogeneous network.

## PROPOSED CROSS-LAYER MODEL

### Wireless Channel Model

To simplify our model, we use the overall cross-layer structure over wired-to-wireless networks (Liu et al, 2004) for end-to-end connection between a server (video source) and a single wireless client (destination). The layer structure of the system under consideration and the processing units at each layer are shown in Figure 3. Hence, we consider a wireless link model across the physical (hardware-radio), data link, and transport layers for each wireless client, which enables us to analytically derive the desired QoS metrics in terms of BER, packet loss and throughput for each wireless client.



**Figure 3.** A cross-layer structure over heterogeneous wired-to-wireless Network

Let  $G$  be the multicast group size and the feedbacks for client  $g$  ( $1 \leq g \leq G$ ) are in terms of the estimated end-to-end available bit rate  $B_g$ , and packet drop rate  $P_{l,g}$  for wired line clients (due to permanently missing sequence numbers of the packets), and for wireless clients, the bit error rate of wireless hop  $p_{e,g}$  (estimated after accounting or limited Automatic Repeat reQuest (ARQ) recovery in the wireless hop or by using a Markov process) (Chiasserini and Meo, 2002). To obtain  $p_{e,g}$ , frequent and random bit errors of a simple noisy wireless channel are considered without taking any fast fading effect. We define the physical layer packet loss rate as a function of the bit error probability  $p_{e,g}$  (or  $\gamma_b$  which is being SNR per bit) for a given modulation mode and packet length in  $L$  as in (Yoo et al, 2004),

$$P_{e,g}(\gamma_b, L) \leq 1 - (1 - p_{e,g}(\gamma_b))^L \quad (1)$$

where  $L$  denotes a packet length (in bits), and the inequality in (1) represents the fact that one can recover from bit errors in a packet, due to the coding scheme. Furthermore, we can define at the radio data link layer the maximum throughput (Goodput) of a channel coding as the number of payload bits per second received correctly for a simple modulation scheme such as Binary-Phase Shift Keying (BPSK) (Yoo et al, 2004),

$$G_{phy,g} = \frac{L-C}{L} \mathfrak{R}_b [1 - P_{e,g}(\gamma_b, L)] \quad (2)$$

Assume  $C$  may not only involve error-correction bits, but any *extra bits* which are related to a header of ARQ packet scheme (if ARQ scheme effect is taken into account) (AL-Suhail, 2008). The term  $[1 - P_{e,g}(\gamma_b, L)]$  denotes the packet success rate (PSR) (i.e., the probability of

receiving a packet correctly),  $\mathfrak{R}_b$  is the source bit rate (in bps) excluding FEC code, and  $\gamma_b$  is the channel SNR per bit (in dB) given by (Yoo et al, 2004),

$$\gamma_b = E_b/N_o = P/(N_o \mathfrak{R}_b) \quad (3)$$

where  $E_b$ ,  $N_o$ , and  $P$  represent the bit energy, the one-sided noise power spectral density, and the received power respectively. 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. Since  $p_{e,g}$  in AWGN channel decays exponentially as  $\gamma_b$  increases, the probability of bit error can be given by (Yoo et al, 2004),

$$p_{e,g}(\gamma_b) = Q\left(\sqrt{2 \gamma_b}\right), \quad (4)$$

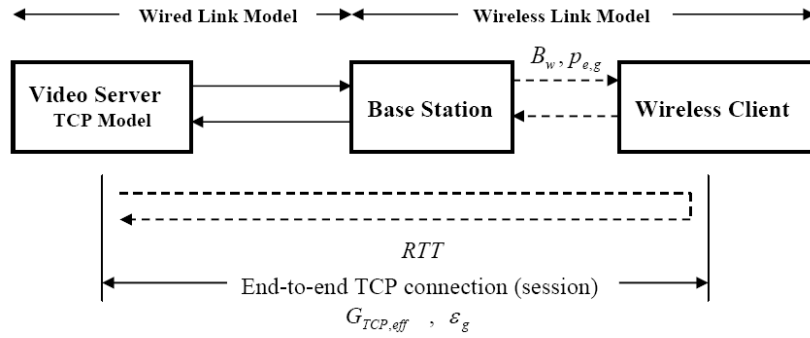
$Q(\cdot)$  is Gaussian cumulative distribution function.

### Adaptive TCP Throughput Formula

Despite the complex behavior of TCP due to its various mechanisms such as slow start, congestion, timeout, etc., it has been shown in (Chen and Zakhor, 2006) 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  $(1 - \varepsilon_g)$

$$G_{TCP,g} \approx \min \left( W_{\max}, \frac{k \cdot S}{\sqrt{\varepsilon_g}} \right) \frac{(1 - \varepsilon_g)}{RTT}, \quad (5)$$

where  $W_{\max}$  is the maximum congestion window size of the TCP sender,  $\varepsilon_g$  is the residual end-to-end packet loss rate for wireless client,  $S$  is the packet size (MSS),  $k$  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 (Liu et al, 2002, Barman et al, 2004). Since (5) does not account for timeouts, it usually overestimates the connection throughput as loss rate increases. It is reported in (Padhy et al, 2000) that (5) is not accurate for loss rates higher than 5%. Upon equation (5), Figure 4 therefore explains the end-to-end analytical TCP model of wired-to-wireless connection



**Figure 4.** An end-to-end analytical TCP model of wired-wireless connection for each wireless client.

which is required to achieve the optimal performance in transporting video for each wireless client end (Chiasserini and Meo, 2002).

### Optimal FEC Codes

According to Figures 2 - 4, we present a possible end-to-end solution which employs TCP based an adaptive concatenated FEC encoding to provide error-resilient video service for each client over heterogeneous IP network. We use TCP goodput formula in (5) to analyze the tradeoff between the gain of the TCP goodput and the reduction of effective channel bandwidth through the application of FEC codes. We provide a cross-layer algorithm to obtain the optimum FEC codes that maximizes TCP goodput and consequently the resultant video play-out frame rate at end clients.

We first propose concatenated byte-level (*inner*) and packet-level (*outer*) FECs to protect the video layers in case of *non-transcoding* gateway whereas both Reed Solomon (RS) codes are performed at the video server and error correction are *only* done at the end clients (Pei et al, 2004; Liu et al, 2004). Second, link layer agent is assumed in the server, the base station and the mobile host respectively to improve the TCP throughput. That's each wireless client effectively receives the video sequence with the bit rate allocated including FEC encoding (done at video server) equal the minimum end-to-end bit rate (done at gateway), where the gateway (at base station) *does not* recover any dropped packets by the packet-level FEC or even pad the video packets with byte-level FEC parity. Note that with this system, byte-level FEC does not effectively help those wired clients (where packet drops occur) in improving their error resilience capability. Thus the *only* concern at the video server is how much error control (FEC codes) should be applied to serve both wireless and wired clients so that their overall temporal quality at frame-level is optimized. However, it is noticed that in many

studies, *e.g.* (Chan et al, 2006; Bajie et al, 2006; Lee et al, 2002; Zorzi et al, 2002), the quality is measured by the aggregate goodput in the system, or equivalently, average goodput of the clients.

In this work, we consider an  $(n, k)$  Reed-Solomon codes to adjust the level of redundancy at byte and packet information. An  $(n - k)$  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 (bits) or packets. Now, to generate the byte-level FEC based on RS code, the encoder processes in symbols, where each symbol consists of  $m$  bits ( $m=8$  in general). Given a packet size  $n_b$  (in byte),  $k_b$  (in byte),  $k_b (\geq 1)$  bytes of source data are packed with  $n_b - k_b$  parity bytes, where  $k_b = n_b, n_b - 2, \dots$ . This is the so-called  $RS(n_b, k_b)$  code, which is able to correct up to symbol errors in a packet, where  $t_b = \lceil (n_b - k_b) / 2 \rceil$ . The packet size  $n_b$  is limited by  $2^m - 1$  symbols; therefore, for  $m=8$ ,  $n_b \leq 255$ .

With every  $k_b$  of these byte encoded video packets, a packet level FEC is then applied to generate  $n_p - k_p$  parity packets to form a block of  $n_p$  packets. Here, the  $i$ th byte of each of the  $k_p$  video packets ( $1 \leq i \leq n_b$ ) is taken out to generate parity bytes. The generated parity bytes are then redistributed as the  $i$ th byte of each  $n_p - k_p$  parity packets corrected up to  $t_p = n_p - k_p$  packet losses.

On the other hand, the server computes the optimal allocation between the video data rate, the packet-level FEC rate (number of packet parity bits per second, *i.e.*, the outer rate,  $\mathfrak{R}_{outer} = k_p / n_p$ ), and the byte-level FEC rate (number of byte parity bits per second, *i.e.*, the inner rate,  $\mathfrak{R}_{inner} = k_b / n_b$ ) for given feedbacks from the end clients. Here, the server has first decide the packet-level and byte-level FEC rates with its transmission rate  $\mathfrak{R}_b$ , including all the redundant bits is equal to the least end-to-end bit rate in the multicast group, *i.e.*,  $\mathfrak{R}_b = \arg \min_g(B_g)$ . Consequently, the resultant video source rate  $G_C$  excluding all the FEC is given as (Pei et al, 2004),

$$G_{C,g} = \mathfrak{R}_b \times \mathfrak{R}_{outer} \times \mathfrak{R}_{inner} \quad (6)$$

Let  $\Gamma_{C,g}$  be the goodput of the  $g$ th client. Therefore, we study the following byte-level and packet-level FEC allocation problem: Given  $n_b$  and  $n_p$ , find the optimal  $k_p$  and  $k_b$  in order to maximize the end-to-end goodput of all clients,

$$\Gamma = \sum_{g=1}^G \Gamma_{C,g} \leq B_w \quad (7)$$

Such that the end-to-end packet loss rate after error correction is no more than certain threshold value,  $\mathcal{E}_{Threshold}$  (say, 1%-3%) for loss rate over all clients. Further,  $\Gamma \leq B_w$  and  $\Gamma_{C,g} \leq B_w$ , where  $B_w$  is being a limited wireless channel capacity. We consider all clients in the system have the same priority or importance (Chen and Zakhor, 2006).

Let us consider a particular client  $g$  and obtain its goodput given  $P_{l,g}$  and  $p_{e,g}$ . In the wireless hop, the symbol error rate is computed from (1) for  $m=8$  bits and  $L= n_b =255$  bytes. Since the RS  $(n_b, k_b)$  code corrects up to  $t_b$  symbol errors, the probability that a random packet cannot recovered by byte-level FEC is given by (Lee et al, 2002),

$$\alpha_g = \sum_{j=t_b+1}^{n_b} \binom{n_b}{j} P_{e,g}^j (1 - P_{e,g})^{n_b-j} \quad (8)$$

Note that for wired clients,  $\alpha_g = 0$  as  $p_{e,g} = 0$  by definition.

In contrary, a packet is correctly received by the client, only if it is not dropped from the overflow or blocking in wired networks with probability  $(1 - P_{l,g})$  and is correctly received through the wireless channel with probability  $(1 - \alpha_g)$ . Hence, we express the packet (segment) loss rate, as in (Liu et al, 2004):

$$p_g = 1 - (1 - P_{l,g})(1 - \alpha_g) \quad (9)$$

Note that the dropped packets may be recovered by the packet-level FEC. Then, the probability that a random packet is permanently “lost”, i.e., the end-to-end packet loss rate *after* error correction is given by,

$$\mathcal{E}_g = \sum_{k=t_p+1}^{n_p} \frac{k}{n_p} \binom{n_p}{k} p_g^k (1 - p_g)^{n_p-k} \quad (10)$$

To reduce the complexity of optimization, we use the same two-step procedures mentioned in (Lee et al, 2002) as follows:

**Step (1)** : Packet-level FEC optimization is used to find the value of  $k_p^*$  so that the residual loss rate over wired network is no more than  $\mathcal{E}_{Threshold}$  ignoring wireless links errors by setting  $\alpha_g = 0$  for all clients. Let  $P_l = \max_g P_{l,g}$  be the maximum packet drop rate for all the

clients. If  $P_l \leq \mathcal{E}_{Threshold}$ , STOP and proceed to next step whereas the packet drop rate is so low that  $k_p^* = n_p$ . Otherwise, for all the clients with  $P_{l,g} > \mathcal{E}_{Threshold}$ , we need to search for the largest  $k_p \leq n_p$  such that  $\mathcal{E}_g \leq \mathcal{E}_{Threshold}$ .

**Step (2)** : Byte-level FEC optimization is followed to find the value of  $k_b^*$ , for given  $k_p^*$  and re-introducing  $\alpha_g$ , that the largest  $k_b \leq n_b$ , such  $\mathcal{E}_g$  in (9) for all the wireless clients are no more than  $\mathcal{E}_{Threshold}$ . Hence, the effective link bandwidth (goodput) of the client  $g$  can be obtained after manipulating the formula (2) to match the model requirements. Then,

$$\Gamma_{C,g}^* = G_{C,g}^* (1 - \mathcal{E}_g) = \mathfrak{R}_b \times \mathfrak{R}_{outer}^* \times \mathfrak{R}_{inner}^* (1 - \mathcal{E}_g) \quad (11)$$

### Optimal TCP-Adaptive FEC over Wireless

According to our model in Figure 4, given the estimate of  $\mathcal{E}_g$  in (10) and a certain value of  $RTT$ , we can compute the available TCP goodput ( $G_{TCP,g}$ ) using (5). Then, to achieve the optimal TCP-adaptive FEC for wireless clients, the real TCP goodput ( $G_{TCP,eff}$ ), i.e., the optimal allocated bandwidth required for TCP protocol, can be accounted for the minimum of the achievable TCP-Friendly goodput ( $G_{TCP,g}$ ) and the effective link bandwidth ( $\Gamma_{C,g}$ ), as in (Liu et al, 2002),

$$G_{TCP,eff}^*(k_p^*, k_b^*, \mathcal{E}_g) = \min(\Gamma_{C,g}(k_p^*, k_b^*, \mathcal{E}_g), G_{TCP,g}(k_p^*, k_b^*, \mathcal{E}_g)) \quad (12)$$

As a result, the optimal RS codes  $(n_p, k_p^*)$  and  $(n_b, k_b^*)$  are the codes that maximizes TCP throughput after error correction at end clients (i.e., TCP Goodput), and are computed as follows (Liu et al, 2004):

$$k_p^*, k_b^*, \mathcal{E}_g^* = \arg \max_{k_p \in n_p, k_b \in n_b, \mathcal{E}_g \in p_g} G_{TCP,eff}(k_p, k_b, \mathcal{E}_g) \quad (13)$$

In point of the fact, the TCP throughput is maximized when the achievable TCP throughput equals the effective channel bandwidth. Ideally each  $k_p$  and  $k_b$  are just the solution to the equation  $\Gamma_{C,g} = G_{TCP,g}$ . Thus the protocol, which is based on the client feedback via the base station (gateway), uses a lookup table at the video server generated a priori to find the code that yields the largest goodput as a function of channel SNR estimate.

## Temporal Video Model

Using a TCP-adaptive FEC scenario, the predicted optimal playable frame rate (PFR) based on maximum effective TCP throughput in (12) can be evaluated as (Wu et al, 2005; AL-Suhail, 2008),

$$R_{g,eff}^* = G \cdot W_I \cdot [1 + \chi_P + N_{BP} \cdot W_B (\chi_P + W_I \cdot W_P^{N_P})] \quad (14)$$

where,

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

$$G_{GoP}^* = \frac{G_{TCP,eff}^* / L}{S_I + N_P S_P + N_B S_B}. \quad (15)$$

$W_i$  stands for the successful transmission probability of the  $i$ -th frame type (I, P, and B) in a GoP pattern taking into account the end-to-end packet loss rate after error correction given in (10).  $S_i$  denotes packet size of the  $i$ -th frame type. In our analysis, packet length must be fixed at 255 bytes in case of byte-level FEC,  $G_{TCP,eff}^*$  is defined as the effective network throughput received at the client in (bps) or in other words, the optimal allocated bandwidth required for TCP protocol under the constraint of (12), and  $G_{GoP}^*$  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), respectively.

## NUMERICAL RESULTS

We assume clients profile of loss rates used for non-transcoding gateway as in Table (1) (Lee et al, 2002). As a reference, we fix a typical set of parameters as: threshold error  $\varepsilon_{Threshold} = 1\%$ ,  $n_b = 255$  byte,  $n_p = 40$  packet, and  $\mathfrak{R}_b = 100$  kbps. We consider also a baseline system of  $G = 10$  clients, with half of them are being wireless and the remaining are wired with uniformly distributed with mean  $P_{l,g} = 2\%$  and  $p_{e,g} = 10^{-4}$  (i.e., average SNR,  $\gamma_b = 8.4$  (dB) for a simple random error of AWGN channel), respectively. Furthermore, Table (2) contains of wireless network and GoP parameters used in simulation. Optimal FEC allocation has been conducted for given these parameters using the two-step procedures. For example, the resultant predicted PFR of (14) is 9.32 [fps] in case of client **C#3** using the optimization procedure defined in Steps 1 and 2, respectively. The corresponding optimal values of the design parameters are



achieved as follows: optimal TCP goodput ( $G_{TCP,eff}$ ) equals 93 kbps, for the obtained values of  $k_p^* = 38$ ,  $k_b^* = 249$  (251), and  $\varepsilon_g^* = 3.7 \times 10^{-3}$  ( $8.57 \times 10^{-4}$ ) at a minimum given  $RTT$  of 168 ms in 1xRTT CDMA network (Chen and Zakhor, 2006; AL-Suhail, 2008). Note that for  $p_{e,g} = 10^{-4}$ , the byte-level FEC occurs in packet error with probability of 0.1845 using (1), whereas  $L = m.n_b$ . With this packet-loss rate, only few parity bytes (about 4-6) are enough to bring this error rate down to a low level given by  $\varepsilon_{Threshold}$  (Lee et al, 2002).

**Table (1).** Clients' profile used for proposed cross-layer model

Client #	$P_{l,g}$ (%)	$p_{e,g}$ ( $10^{-4}$ ),	Error Type
<b>C#1</b> wireless	2.7698	1.0134,	random error AWGN
<b>C#2</b> wireless	2.4790	0.8594,	random error AWGN
<b>C#3</b> wireless	2.0572	0.9993,	random error AWGN
<b>C#4</b> wireless	1.8248	1.3363,	random error AWGN
<b>C#5</b> wireless	1.7179	0.5460,	random error AWGN
<b>C#6</b> wired	1.1049		0.0
<b>C#7</b> wired	1.3341		0.0
<b>C#8</b> wired	2.1079		0.0
<b>C#9</b> wired	2.4529		0.0
<b>C#10</b> wired	2.7578		0.0

**Table (2).** Wireless network and GoP pattern parameters used in simulation

Parameter	Wireless Network Design Parameters
$RTT$	168 [ms]
$k$	1.22
$B_w$	1Mbps (1xRTT CDMA)
$L_{max}$	255 byte*
Modulation	BPSK (upload/download)
$\gamma_b$ (AWGN)	5...10 [dB] Channel SNR/bit
	<b>GoP Pattern Design parameters</b>
$F_o$	30 [fps] reference frame rate at video server
GoP(2,3)	<i>I-BBB-P-BBB-P-BBB</i>
$(S_I, S_P, S_B)$	Typical values (25,8,1), (10,3,1), (7,2,1) [packet]

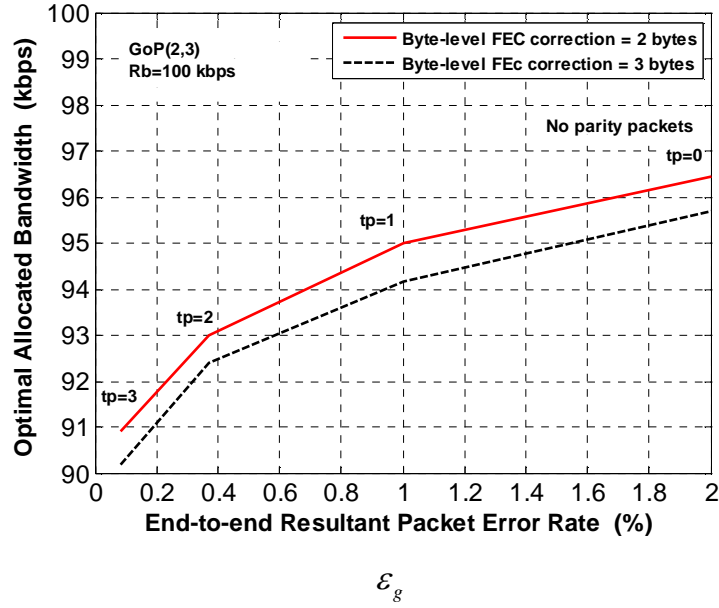
\* We assume that a TCP packet size (MSS) is equal to the maximum length of packet (block) at the data-link layer (Barman et al, 2004).

Effectively, it indicates the efficiency of byte-level FEC. In fact, once  $k_p^*$  is increased, the packet-level error correction efficiency decreases, and hence a lower  $k_b^*$  (i.e., stronger byte-level correction efficiency) is needed to adapt the overall bandwidth. Hence, the optimal allocated bandwidth for TCP in (12) is analytically affected by the two factors: the optimal end-to-end packet loss rate  $\varepsilon_g^*$  (i.e., optimal bandwidth required in (12) decreases if  $\varepsilon_g^*$  is

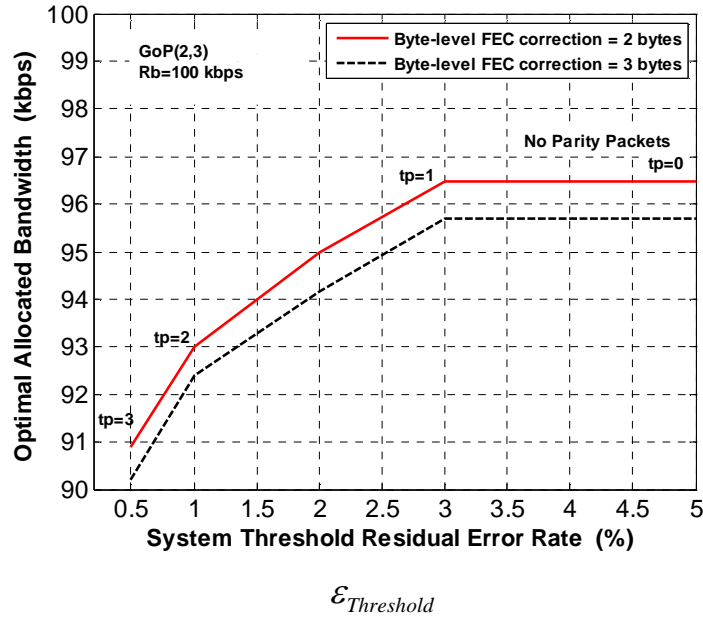
increased), and the optimal parameters  $k_p^*$  and  $k_b^*$  (i.e., optimal bandwidth required in (12) increases if both parameters are increased). Accordingly, the effective TCP throughput can be evaluated through the optimal allocated bandwidth that is required to compensate the high values of residual packet loss rate when low values of FEC parity symbols (at byte-level as well as packet-level) are allocated for video bitstream. Figure 5 illustrates the optimal allocated bandwidth vs. the end-to-end residual packet loss rate for various values of FEC codes. Meanwhile, Figure 6 displays the corresponding system threshold required of a residual error rate vs. optimal allocated bandwidth for some significant values of packet-level FEC. It is clearly noticed that when  $\varepsilon_{Threshold}$  increases then the resultant  $\varepsilon_g^*$  increases and consequently the optimal allocated bandwidth required in this case considerably increases according to our model in (12). It means that we should choose only a minimum required bit rate as an optimal allocated bandwidth for TCP protocol for each wireless client. In contrast, the throughput of each client will be limited by the upper bound of the video source data  $\mathfrak{R}_b = 100$  kbps.

On the other hand, it is found that when the optimal  $t_p^*$  is increased the end-to-end optimal bandwidth required is clearly reduced for a given byte-level FEC codes. The results obtained are compatible with corresponding ones in (Liu et al, 2004), but with different network conditions. In our model, Figure 7 reveals the optimal PFR vs. the system threshold of residual error rate under different packet size for each  $i$ -th frame type of GoP. Specifically, PFR in (14) depends on two design parameters:  $\varepsilon_g^*$  and the optimal allocated bandwidth of TCP protocol. The later parameter can basically depend indirectly on  $\varepsilon_g^*$  in case of  $\Gamma_{C,g}$  is being greater than  $G_{TCP,g}$  in (12).

Since the optimal allocated bandwidth based TCP is limited by the upper bound of video source rate,  $\mathfrak{R}_b$ , it is noticed that for a value of  $k_p^* = 38$ , there is no significant change in PFR when  $t_b^*$  is increased from 2 to 3 bytes (Lee et al, 2002). In our model, given feedback of clients the video server needs to adapt the resultant PFR at the client by choosing either the certain threshold value  $\varepsilon_{Threshold}$  or appropriate packet size for each frame type in GoP pattern, or both. Therefore, it is noticed clearly that the play-out frame rate outperforms as far as the frame's packet size is shortened at the video server in order to compensate any packet drop in each frame of GoP. An example is given in Table (3). The results obtained are compared depending on the maximum throughput of BCH channel coding in (AL-Suhail, 2008). The effective PFR can significantly rise to achieve 20.52 [fps] and 26.16 [fps] at wireless client



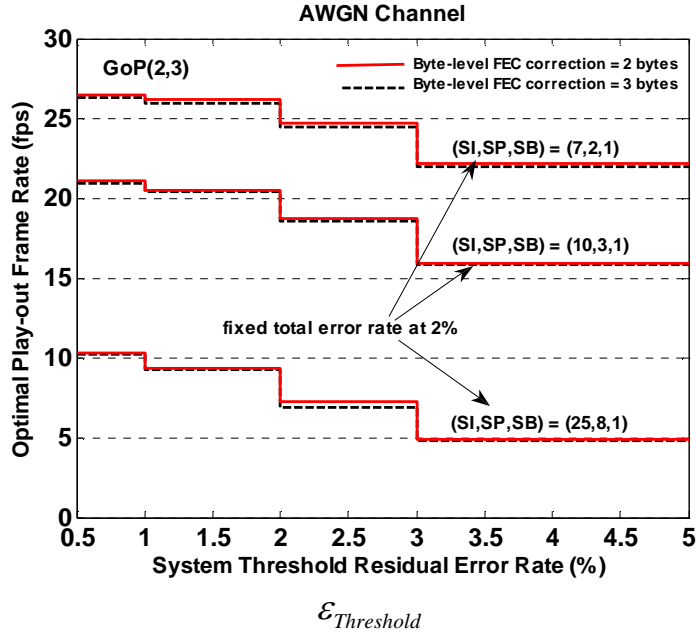
**Figure. 5** Optimal allocated bandwidth vs. the end-to-end resultant packet error rate.  $P_{l,g} = 2\%$  and  $p_{e,g} = 10^{-4}$ ,  $R_b = 100$  kbps,  $n_b = 255$ , and  $n_p = 40$  for various FEC codes.



**Figure. 6** Optimal allocated bandwidth vs. the system's threshold value of residual error rate.  $P_{l,g} = 2\%$  and  $p_{e,g} = 10^{-4}$ ,  $R_b = 100$  kbps,  $n_b = 255$ , and  $n_p = 40$  for various FEC codes.

**C#3** for different packet size settings (generated at video server) of I, P, and B frames, such as (10, 3, 1) and (7, 2, 1), respectively. Thus we can conclude that as the packet size of each  $i$ th frame type (I, P, and B) decreases the play-out frame rate considerably increases at the client ends in order to compensate the frame dropping process in our model compared to example of approach in (AL-Suhail, 2008). However, the system threshold value of residual error rate has a significant effect on video quality. As this certain value (at video server)

increases to be greater than 3%, the quality degradation will be constant due to the fixed achievable value of residual packet error of (10) at 2%. Moreover, extra TCP goodput results can also be obtained when the reference video source rate is considered to 160 kbps or 200 kbps for UMTS networks.



**Figure. 7** Optimal play-out frame rate vs. the system's threshold value of residual error rate.  $P_{l,g} = 2\%$  and  $p_{e,g} = 10^{-4}$ ,  $\mathfrak{R}_b = 100$  kbps,  $n_b = 255$ , and  $n_p = 40$  for various packet frame size of GoP.

**Table (3).** A comparative example of MPEG-4 video transport under different network conditions

Approach	Channel State	Error Type*	FEC Parity, Correction Symbols	Residual PLR after error - correction (%)	Optimal TCP Bandwidth (kbps)	$S_I, S_P, S_B$ (packets)	PFR (fps)
Hybrid FEC and ARQ-Based TFRC (AL-Suhail, 2008)	C1 8.40 dB (wireless link)	$1 \times 10^{-4}$ Random Error AWGN	9 ( $t=1$ ) BCH at data link layer	0.126	80.11	25,8,3	26.13
	C2 7.35 dB (wireless link)	$5 \times 10^{-4}$ Random Error AWGN	18 ( $t=2$ ) BCH at data link layer	0.228	71.08	25,8,3	22.13
Proposed Cross-layer based byte-level and Packet-level FECs	C# 3.8dB-8.4dB (wired & wireless links)	$p_{e,g} = 1 \times 10^{-4}$ , Random Error AWGN $P_{l,g} = 2\%$ , Congestion Error	2 ( $t_p = 2$ packets)	0.37-0.0857	93.0 (on average)	25,8,1	9.32
			4-6 ( $t_b = 2-3$ bytes) (RS coding)			10,3,1	20.52
						7,2,1	26.16

\* Error Type= Random Error over AWGN channel, GOP(2,3), RTT=168 ms.

In Table (3), it is also noticed that our proposed approach clearly introduces reasonable performance compared to the cross layer design in (Liu et al, 2004). The optimal values of the residual packet loss rate after correction (PLR) can achieve the values of 0.0857%-0.37% to provide optimal PFR; meanwhile the optimal TCP throughput improvement by using cross-layer design in (Liu et al, 2004) under different channel conditions and effects of design parameters illustrates that the resultant PLR values are no less than 0.3%-0.6%.

As a result, we can conclude that compared to the other related work (Chan et al, 2006; Lo et al, 2005; Lee et al, 2002; Liu et al, 2004; and AL-Suhail, 2008), one of the key features of this cross-layer model is that it uses a formula-based approach to analytically derive the optimal adaptive FEC that maximizes end-to-end TCP throughput and consequently multicasting the optimal video quality at the clients. The required modifications to implement TCP adaptive FEC include some link layer operations at the video server, base station and mobile host. Since the modified link layer operations are transparent to the TCP at the end clients, the end-to-end semantics of TCP is preserved.

## CONCLUSION

In this paper, we presented a new cross-layer model to achieve the optimal TCP throughput for video multicasting over heterogeneous networks using adaptive forward-error correction (FEC). The model integrates the TCP throughput at the transport layer with link layer error control over wireless links. A model based TCP goodput formula is combined with adaptive FEC at byte and packet levels to select the optimal code that maximizes TCP goodput for AWGN wireless channel, and consequently multicasting the optimal video quality at end clients. The results show a good video quality that can be achieved when a maximum TCP throughput is reached at appropriate system settings for the threshold residual error rate and frames size of group pattern (GoP) of MPEG-4 video. For further work in this paper the approach can be developed to involve adaptive modulation and channel coding for more robust video transport in multi-path fading channels in wireless links.

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