QoS Provisioning Using Cross-Layer Design

Tahmid A. Quazi, *Student Member IEEE*, Hongjun Xu, *Member IEEE* and Fambirai Takawira, *Member IEEE*.

Abstract— Guaranteeing QoS is critical to the functioning of multimedia applications over mobile wireless networks. However, currently the challenge of QoS provisioning is being faced using the legacy layered protocol architecture where each layer provides a separate, independent solution, with its own optimized adaptation and protection mechanisms. Cross-layer design has been proposed as a methodology to extend that paradigm in wireless links where there is interdependence between the layers and hence opportunity for information sharing. Recently Cross-layer adaptation mechanisms have been proposed that attempt to solve the QoS provisioning problem. However, most of these mechanisms only use the lower (physical and data link) layers and the possibility of using higher protocol layers remain unexplored. As a result restrictions are placed on the system which introduces functional and efficiency limitations. In this paper we highlight one such limitation, namely the inability to insert more than one class of traffic in a physical layer frame. We then propose a physical and application layer Cross-layer adaptation mechanism that overcomes this. The performance results of the scheme shows that the Cross-layer mechanism can be efficiently applied for the purpose of QoS provisioning in wireless links.

Index Terms—Multimedia traffic, Quality of Service (QoS), Cross-layer design, adaptive transmission.

I. INTRODUCTION

PROVIDING guarantees in the Quality of Service (QoS) has become essential to the operation of today's multimedia wireless networks. However this poses quite a challenge due to the variable nature of wireless links and the diverse QoS requirements of different applications including voice, video and data. In the current paradigm of the layered protocol stack each layer of the communications stack works independently to provide a solution to these challenges. The benefits of dynamic adaptation to system conditions has been accepted, but the true potential of optimized adaptation is lost if the layers operate in an individual fashion, ignoring possible interdependencies between them. Cross-layer design (CLD) provides mechanisms in which these interdependencies are exploited to achieve challenging objectives such providing QoS guarantees over wireless links.

One of the earliest contributions in the research area of Cross-layer design was to increase spectral efficiency by adapting certain parameters of the transmitted signal to match the wireless channel conditions. Physical and data link layers were combined to allow the adaptive variation of transmission power, symbol rate, coding rate, constellation size or any combination of these parameters. The work presented in [1] and [2] showed that efficient bandwidth utilization for a prescribed error performance at the physical layer can be accomplished using adaptive modulation and coding (AMC) algorithms. Recently Liu et al [3], [4] used an AMC scheme to provide QoS support in wireless networks. The objective of the AMC module in such a system is to maximize the data rate by adjusting the transmission modes i.e. varying the modulation and coding schemes to the channel variations while maintaining a prescribed PER (P_0). The limitation, however, of such a method in the TDM/TDMA system is that once a target PER has been chosen (depending on the requirement of the traffic type being transmitted) for a frame, then only that type of traffic can be inserted into the frame. If P_0 has been set for voice traffic then, although the target for voice is met, it will not be sufficiently low for video or data, if such traffic were to be inserted into the frame. If, however, the target has been set for data traffic the AMC system will be inefficient for voice and video traffic. The efficiency of the resource allocation module of the system would be noticeably higher if it was able to insert more than one type of traffic in a frame and it had a finer control over the allocation of resources for the different types of traffic being transmitted. In order to achieve this, we propose a Cross-layer system where the AMC module is combined with an adaptive coding mechanism at the application layer to guarantee QoS performance in terms of loss rate for voice, video and data traffic. The physical layer guarantees a fixed error target for a frame and the application layer coder applies the necessary error protection depending on the type of traffic being inserted in the frame in order to meet the QoS frame error target.

QoS provisioning using Cross-layer design has been proposed by combining the physical and Data Link layers of the protocol stack [3], [4]. In these schemes multiuser scheduling at the MAC layer is employed in conjunction with the AMC module at physical layer. The limitation of such a system has been mentioned in the previous paragraph. The work in [5] has a similar objective to that proposed in this paper, namely to allow the transmission of different classes of traffic in the same frame (or more generally in a unit of transmission), however theirs is a CDMA setting. The concept of combining the physical and the application layer for error protection has been applied in the area of video transmission over wireless links [6], [7]. The physical layer parameters are adapted to the different channel conditions while an application layer FEC provides an additional error control strategy. The novelty of the

The authors can be reached at quazit@ukzn.ac.za, xuh@ukzn.ac.za ftakaw@ukzn.ac.za

application layer FEC for a video transmission is that it can offer the flexibility of unequal or selective error protection. A multiple description source encoder splits a video stream into multiple bit streams or descriptions, and each sub-stream has a different priority. The priority is based on the level of fidelity with which each sub-stream describes the original steam. A higher level of FEC protection is then applied to a high priority bit-stream over one with a lower priority.

The system proposed in this paper applies the unequal error protection concept to the three classes of traffic to be transmitted over the wireless link. Data traffic has the highest priority in terms of QoS error protection and receives the highest level of protection while voice traffic, having lowest priority, receives the lowest level of protection. The remainder of the paper is organized as follows: the system model is described in Section II. Section III describes the design of system algorithm. Performance evaluation is detailed in Section IV and the paper is concluded with future work in Section V.

II. System Model

A. System Description

The system under consideration has multiple users or nodes connected to a central unit which could be a Base Station or a backbone gateway for a WLAN or ad hoc network. Each user/node is connected to the central unit over the wireless channel using time-division multiplexing/time-division multiple-access (TDM/TDMA). Only the downlink is considered in this system, with the assumption that the results for the uplink would not be dissimilar.

The wireless link between two nodes is shown in Figure 1 (A modification of that presented in [4]). The application layer contains the adaptive source coder that varies its coding parameters depending on the channel conditions. A buffer (queue) is implemented at the central node for each user and operates in a first-in-first-out (FIFO) mode. The AMC module follows and precedes the buffer at the transmitter and receiver respectively. The AMC controller is implemented at the central node (transmitter) and an AMC channel estimator at each user (receiver).

The traffic (voice/video/data) streams are encoded by the coder at the application layer to form code blocks. These code blocks are passed to the data link layer as packets. The packets are stored in the buffer waiting to be served by the AMC module of the physical layer. After the AMC module, the symbols are packaged into a frame which is transmitted over the wireless channel. The AMC module allows each user to have multiple transmission modes, each of which represents a pair of a specific modulation format and a FEC code. Such transmission modes are implemented in systems operating the HIPERLAN/2 and IEEE 802.11a protocol standards. Channel estimation is done at the receiver and the results are fed back to the transmitter. This information is fed to both the AMC controller and the adaptive source coder, and they adjust their parameters accordingly. The CSI is used by the AMC controller to determine a modulation-coding pair (or mode) and by the source coder to determine a coding rate.

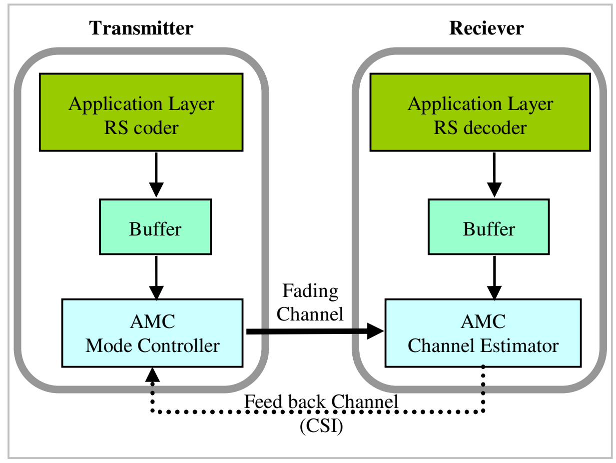

Figure 1 Wireless link from the central node to a user

B. Packet Structure at various layers

- The application layer: The voice, video or data packets are encoded into RS code blocks. The encoder takes *k* data symbols of *s* bits each and adds parity symbols to make an *n* symbol code word/block.
- The data link layer: Each packet contains a fixed number of bits (N_b) and this includes the packet header, the payload from the application layer coder and CRC bits. Each packet is mapped to a symbol block containing N_b / R_n symbols where R_n is the rate for mode *n*.
- The physical layer: Each frame contains a fixed number of symbols (N_s). The symbol rate is fixed and so the frame duration (T_f) remains constant. With TDM, the frame is divided into $N_c + N_d$ time slots, where each time slot contains a fixed number of N_b / R_1 symbols [5]. Given a transmission mode *n*, each time slot will then contain R_n / R_1 packets. For example a time slot will contain $R_1 / R_1 = 1$ packet with mode *n* = 1, $R_2 / R_1 = 2$ packets with mode *n* = 2 etc. The N_c time slots in the frame hold the control information and pilots while the N_d slots convey the data being transmitted. The data time slots are scheduled to different users using TDMA dynamically. The aim of this work is to allow various class of traffic to be transmitted to one user or many users in the same frame.

C. Operating Assumptions

The channel model assumed for this system is a block fading model where the channel between users is frequency flat and invariant for the duration of one frame, but varying from frame to frame. Such a model is suitable for slowly varying wireless channels [4]. The adaptive schemes at the physical and application layer are then adjusted on a frame-by-frame basis. Perfect channel side information (CSI) is available at the receiver which is derived from training-based channel estimation. The feedback channel is assumed to be error free and instantaneous, thus the determined SNR value is fed back to the transmitter without error and latency.

III. System Design

A. Physical layer

The objective of the adaptive algorithm at the physical layer is to maximize the data rate, while maintaining the required bit error rate (BER) performance at target value. The transmission modes (no coding is used currently in the system) at the Physical layer are arranged such that the rate increases as the mode index *n* increases. This is shown in Table 1 below.

Table 1 Boundary points for adaptive non-coded modulation

Mode(n)	1	2	3	4	5	6	7
M_n	BPSK	4PSK	8PSK	16QAM	32QAM	64QAM	128QAM
SNR(dB)	6.8	10.3	13.9	17.2	20.4	23.5	26.5
R_n	1	2	3	4	5	6	7

Let N denote the total number of transmission modes available. Assuming constant power, the SNR range is partitioned into N + 1 non-overlapping consecutive fading intervals with the boundary points denoted as $\{\gamma_n\}_{n=0}^{N+1}$

Thus mode *n* and thus constellation size M_n is chosen when: $\gamma \in [\gamma_n, \gamma_{n+1}) \quad for \quad n = 1, 2, ...N$

To avoid deep fades, no payload bits are sent when the SNR is below a certain threshold. Table 1 shows the boundary points for the non-coded modulation scheme for a target BER value of 10-3 (appropriate for voice traffic) using M_n-QAM over an AWGN channel. The expressions to determine the boundary points are derived in [1]. When the switching thresholds (modes) are chosen according to the boundaries, the physical layer will ensure a BER performance that is below the target BER.

B. Application layer

The BER performance target chosen at the physical layer was 10^{-3}. This error target, although sufficient for voice applications, is not low enough for more demanding applications such as video and data. In order to accommodate these types of traffic in the system additional error protection is required. The variable error protection is achieved in this CLD system using Reed-Solomon (RS) codes at the application layer.

The analytical expression for Frame error rate for general RS codes is as follows:

The probability of frame error (P_{fe}) can be expressed as:

$$P_{fe} = \sum_{K=T+1}^{N} \binom{N}{K} (1 - P_S)^{N-K} P_S^K \qquad (1)$$

where:

N : Number of code symbols per frame

T : Number of code symbols the code can correct

P_S: Probability of code symbol error

K : Number of data symbols

Then P_S can be expressed as

$$P_S = 1 - (1 - P_b)^M \qquad (2)$$

where:

P_b : Bit error probability

M : Number of bits per code symbol.

The frame error rate was evaluated for a 63 and 127 length RS (n,k) code with error correcting capabilities of t = 1, 2, 3 and 4 over an AWGN channel within the range for each mode of the AMC module.

The threshold values for the k value, i.e., the level of protection for each class of traffic, was determined by setting P_{fe} targets for each class. The targets for voice, video and data were 10^{-1}, 10^{-2} and 10^{-4} respectively. These values are similar to that used in [5] which is another Resource Allocation Scheme using Cross-layer design. Thus similar to the adaptation by the AMC module at the Physical layer, the level of protection is varied depending on the type of traffic and the CSI information (SNR value of the channel) fed back to the transmitter

IV. Performance Analysis

The first task in the performance analysis was to determine the theoretical performance of the system using the threshold boundary points and analytical expressions for both the AMC module at the physical layer and the variable error protection system at the application layer. The number of bits for each RS code symbol (M) used was 8 bits. Shortened (255,k) RS codes were employed with a codeword length (N) of 127 for voice traffic and 63 for video and data traffic. The performance of the three classes of traffic were determined separately using the expressions for BER [1] and FER, and can be seen in Figure 2, 3 and 4 for voice, video and data respectively.

The system was then simulated at certain SNR points over an AWGN channel using the Communications toolbox in Matlab. The message or packet length was $N * M$ and was modulated at the physical layer by the schemes shown in Table 1. The figures above show that the there is a tight match between the simulation and theoretical results. More importantly, they show that the performance of the adaptive application and physical layer Cross-layer design system meets the target for all three classes of traffic across the channel SNR range. This implies that the designed system will allow for the insertion of any of the three classes of traffic, in any combination, in a single physical layer frame without the danger of not meeting frame error rate QoS targets, thus fulfilling the purpose of the proposed Cross-layer design system. The overall system benefits, beside the functional one, include the increased efficiency of the resource allocation mechanism which will now have a fine grain control of the available system resources (timeslots) when scheduling the three classes of traffic

These figures above also show the effect of removing the adaptive application layer coder from the system. The graphs showing the performance of the system without the use of the RS coding at the application layer shows the ineffectiveness of only using the adaptive physical layer mechanism in meeting the error rate QoS targets.

V. Conclusion

In this paper we have emphasized the need for Cross-layer design in providing QoS guarantees in multimedia wireless networks. Current CLD resource allocation mechanisms use a static form of allocation where a physical layer frame is designed to hold only one class of traffic. The proposed system facilitates a dynamic resource allocation where more than one or all three traffic classes can be transmitted simultaneously in one frame. This not only has functional uses in providing QoS in a multimedia wireless system, but also makes the resource allocation process more efficient. The predicted result of this is increased data link layer throughput and lower average delays. Proving this will be the aim of future work. We will try and improve the data throughput by optimizing the length of the RS codes in the adaptive application layer coding process. In addition we will investigate the use of alternative, non-ideal channel estimation and feedback mechanisms for the Cross-layer design system.

Acknowledgement

The authors would like to thank the Telkom Center of Excellence at the School of Electrical, Electronic and Computer Engineering, University of KwaZulu-Natal for their continuous support for this research work.

References

[1] M. S. Alouini and A. J. Goldsmith, "Adaptive modulation over Nakagami fading channels,"Kluwer Journal on Wireless Communications, vol 13, no. 1-2, pp 119-143,May 2000.
[2] K. J. Hole, H. Holm, and G. E. Oien, "Adaptive multidimensional coded modulation over flat fading channels," IEEE J. Sel. Areas Commun., vol.18, no. 7, pp. 1153–1158, Jul. 2000.
[3] Q. Liu, X Wang and G. B Giannakis, "A Cross-Layer Scheduling Algorithm With QoS Support in Wireless Networks", IEEE Transactions on Vehicular Technology, vol 55, no 3, pp. 839-847 May 2006.
[4] Q. Liu, S Zhou and G. B Giannakis, "Cross-Layer Scheduling With Prescribed QoS Guarantees in Adaptive Wireless Networks", IEEE Journal on Selected Areas in Communications, vol. 23, no 5, pp. 1056-1065, May 2005.
[5] S. Ci, M. Guizani and G. Benbrahim, "A Dynamic Resource Allocation Scheme for Delay-Constrained Multimedia Services in CDMA 1xEV-DV Forward Link," IEEE JSAC Special Issue on "Next Generation CDMA Technologies,", Vo. 24, No. 1, pp. 46-53, January 2006.
[6] Y. Chan, P. Cosman, and L. B. Milstein, ``A Cross-Layer Diversity Technique for Multicarrier OFDM Multimedia Networks," *IEEE Transactions on Image Processing*, vol. 15, no. 4, pp. 833-845, April. 2006.
[7] S. Krishnamachari, M. VanderSchaar, S. Choi, and X. Xu, "Video Streaming over Wireless LANs: A Cross-Layer Approach", in *Proc. IEEE Packet Video '2003*, Nantes, France, April 2003.

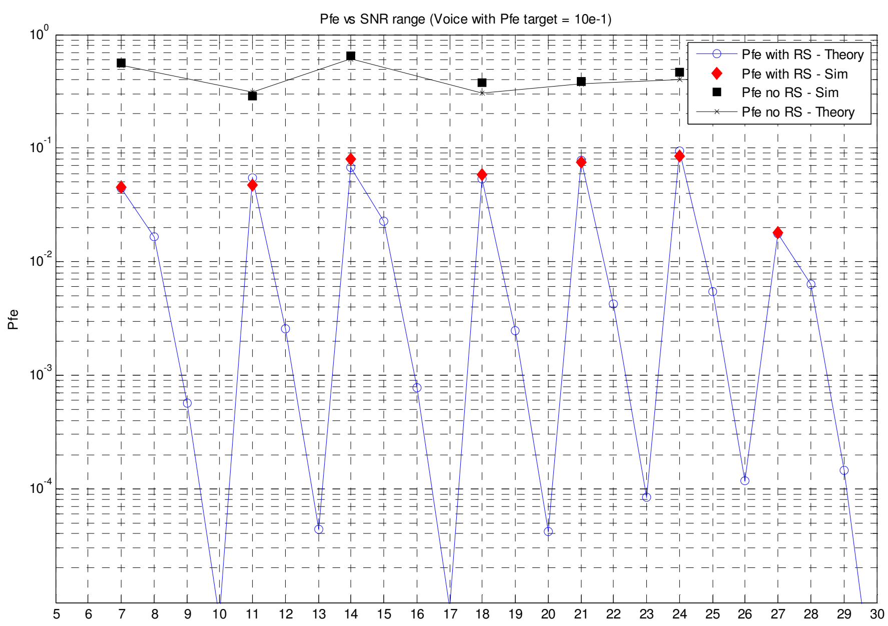

Figure 2 Frame Error Rate performance for Voice traffic

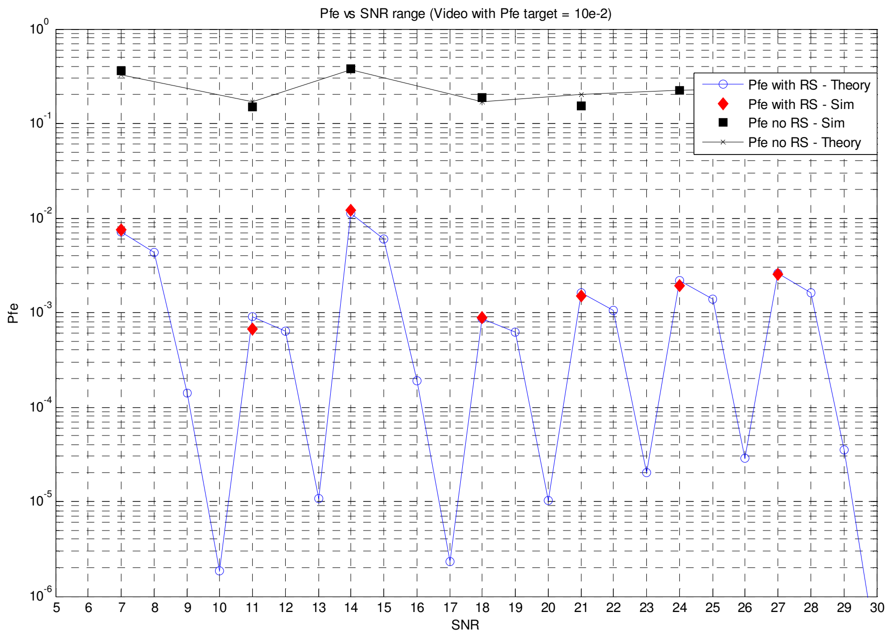

Figure 3 Frame Error Rate performance for Video traffic

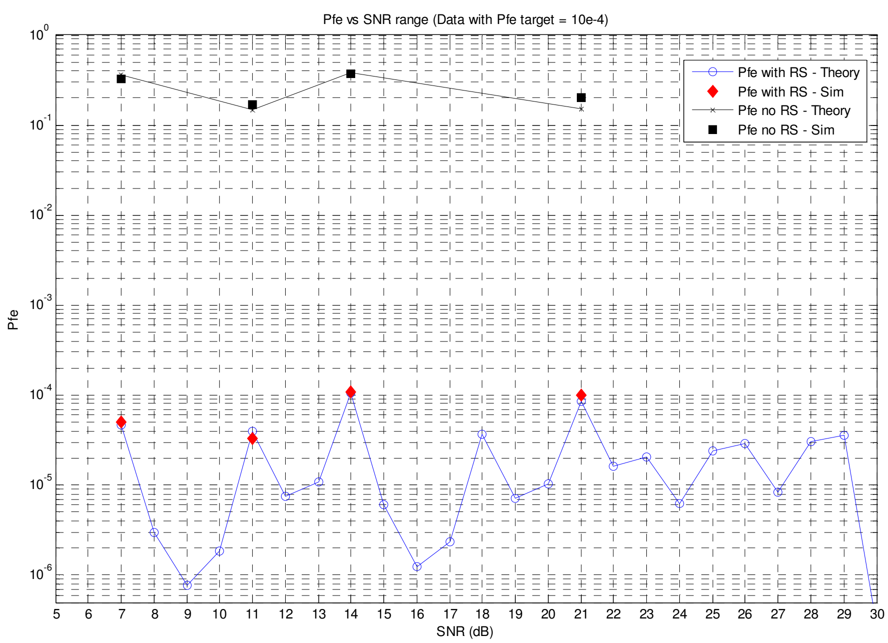

Figure 4 Frame Error Rate performance for Data traffic

Tahmid A. Quazi completed his BSc. and MSc. Engineering (Electronic) degree in 2001 and 2003 respectively, in the School of Electrical, Electronic and Computer Engineering at the University of Natal (Durban). He is currently doing his PhD degree, focusing on Cross-layer design in wireless multimedia networks.

Dr. Hong-Jun Xu received the BSc degree in 1984 from the University of Guilin Technology and the MSc degree from the Institute of Telecontrol and Telemeasure in Shi Jian Zhuang, 1989, and the PhD degree from the Beijing University of Aeronautics and Astronautics in 1995. His research interests are in the area of digital and wireless communications and digital systems.

Professor Fambirai Takawira is head of the School of Electrical, Electronic and Computer Engineering at the University of KwaZulu-Natal. His research interests are in the general areas of adaptive signal processing, digital communications and data networks.