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Robust Real-time Multimedia Transmissions in Wireless Ad hoc Networks

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***Abstract*— To support the quality of service in wireless multimedia communications, wireless systems should work well in typical wireless environments characterized by the path loss of the signals, multipath fading, interference to adjacent channels, and random errors. A variety of researches in the application-layer have been proposed to adapt the video encoder to the link quality by compressing the data with different rates. However, these approaches introduce additional overhead and are inefficient when the round-trip delays are long. This paper proposes an algorithm to prevent packet losses and skipped frames by using optimal fragmentation technique with rate adaptation. Using an adaptive SNR estimator, the sender estimates the SNR of the receiver, and shapes arbitrary sized packets into optimal length packets with rate adaptation. Through rigorous analysis and extensive experiments with an implemented test-bed, we show that the algorithm significantly enhances the visual quality of real-time multimedia streaming in noisy ad hoc networks. The experimental results reinforce the fact that the algorithm is a realistic approach that can be deployed without modifying existing standards.**

Keywords-Fragmentation, wireless multimedia transmission, real-time video streaming

I. INTRODUCTION

Real-time multimedia streaming in wireless ad hoc networks is a challenging problem due to delays and packet losses caused by time varying wireless channels and link failures resulted from dynamic behavior of mobile users. The non-real-time playback of video files is possible using TCP-IP for the transmission of video files. However, delay sensitive real-time video streaming is transmitted using the UDP. Although, UDP provides a time-bounded service, the delivery of the video packets is not guaranteed. To ensure successful transmission of the packets, strong error resilience and concealment in the video codec is inevitable. This adaptation at the application layer requires that the encoder needs to know the quality of the link based on feedback information from the decoder. In ad hoc networks, as longer end-to-end delays and link failures are common, this approach is limited to be deployed in reality.

Link adaptation is another approach that has been studied to provide the quality of service in wireless networks by adapting the frame size and PHY modulation to match the time varying wireless channels. The architecture for adapting frame length to a time varying channel is proposed in [4]. It exploits the effect of bit error rate and frame length on throughput in wireless network. Simple backoff based frame length adaptation [5] and tuning fragment size to fit in a dwell time in the frequency hopping system [6] [7] are proposed. However, these approaches are for a general MAC protocol. In [8] [9] and [10], a link adaptation strategy is studied to select the optimal combinations of the 802.11 PHY mode and the fragment size to achieve the best throughput performance for different SNR conditions. Although the scheme achieves some degree of optimization, it excludes the effect of collisions and thus applicability is limited to a single user case. J. Yin et al models the effect of the contentions among users, the collisions, and the random errors at the receiver in [11]. In the analysis, the optimum packet size is computed in an error prone channel. However, how arbitrary sized packets can be fit into the optimum sized packets remains unexplained. Rate adaptive protocol with dynamic fragmentation [12] combines fragmentation with existing rate adaptation schemes. Basic operation of the protocol is similar to RBAR [2] in that it exchanges channel information using modified RTS/CTS packets. Although, this algorithm achieves much better throughput compared to RBAR [2], it has the same problems of rate adaptation approaches. In addition, the typical WLAN channel is slow fading, and the channel coherent time is long enough to hold multiple packet transmissions. Therefore, actual performance benefit of this algorithm in a typical indoor WLAN is limited, while incurring control packet overheads and protocol change in existing deployed 802.11 systems.

In this paper, a dynamic optimal fragmentation with rate adaptation technique is presented for real-time multimedia transmission in CSMA/CA ad hoc networks. Most of the existing solutions described earlier in this section need unrealistic assumptions and modifications of the protocols. Therefore the results can be shown only through the simulations that are highly dependent on parameters and environments of the simulations. The main contribution of this research is not to present a complicated theoretical algorithm but to provide a feasible practical solution with low complexity such that the algorithm can be implemented and the results can be compared in real-world scenarios. The algorithm adapts rate and packet length to reduce packet

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losses and skipped frames, while maximizing goodput for given network conditions in time varying channels. To estimate the SNR of the receiver, an adaptive estimator that uses on-demand UDP messages with low overheads is designed to avoid modification of existing protocols. The performance of the proposed approach is verified by an implemented test-bed consisting of four mobile stations in a multi-hop ad hoc network.

The remainder of this paper is structured as follows. Section II presents a detailed analysis of fragmentation, goodput, and delay of the CSMA/CA MAC. The system design elements are discussed in Section III with the rate selection algorithm, adaptive channel estimator, impact of imperfect channel estimation, and implementation issues. In Section IV, we discuss our wireless test-bed, the experiments performed in a multi-hop ad hoc network, and the results. Finally, we state our conclusions in Section V.

II. CSMA/CA ANALYSIS

A. CSMA/CA MAC Overview

The fundamental access method in the IEEE 802.11 is a carrier sensing multiple access/collision avoidance (CSMA/CA) mechanism to avoid collisions in the medium while users contend to access the channel. If contention free access is required, point coordination function (PCF) built on the top of the DCF is provided.

In DCF, stations sense whether the medium is idle or occupied by other stations before sending data. If the medium is idle for a DCF inter-frame space (DIFS) interval, the station decreases its backoff timer, which is randomly selected at the first attempt of the transmission. When the backoff timer expires, it transmits data. If the transmission is successful, the station resets the backoff timer and chooses a new time slot in the contention window. A station that has failed the first round of transmission should exponentially backoff the contention window and retry later when the medium is idle. This exponential backoff of the contention window will repeat until it reaches its maximum of 1023 slots for the 802.11b direct sequence spread spectrum (DSSS) physical layer. Then it remains there unless the timer is reset by a successful transmission, or discarded by the retry counter. Because DCF operates without a central coordinator, the medium access control is done independently. This typical exponential backoff ensures the stability of the network and guarantees long-term fairness even in the maximum saturated traffic with many contending stations in the same BSS.

B. Fragmentation and Goodput Analysis

Packet length adaptation is another approach that has been studied to increase the throughput by changing the frame size to match the time varying wireless channels. If transmitters break messages into smaller fragments for sequential transmission, each shorter duration fragment has a better chance of escaping burst interference and decreases fragment loss rate. This simple technique also reduces the need for retransmission in many cases, and can be used to reshape arbitrary sized packets into optimal length packets to improve wireless network performance. However, it incurs overhead on every fragment rather than every frame, thereby reducing the aggregate throughout and realizable peak throughput.

To find optimum fragment and adapt it dynamically in noisy CSMA/CA mobile environments, we consider DCF under saturated traffic conditions with an emphasis on implementation feasibility. In [13], a complicated Markov model is presented to employ exponential backoff in the assumption of ideal channel conditions. To implement dynamic optimal fragmentation with rate adaptation with low complexity, we use a practical model suggested in [14] and [15]. This model is a simple but comprehensive analytical model that considers fragmentation overheads and time intervals. Even though the methodology to derive the formula is similar to [11], the actual computation of the optimal sizes is different due to fragmentation overheads and time components. Furthermore, packet lengths and transmission rates are changing dynamically while [11] is a simple calculation of the packet lengths in static environments.

The key assumption of the model is that the unsuccessful transmission probability, which is the resultant from collisions or corrupted random bits, is a constant and independent probability seen by a packet being transmitted in a randomly chosen time slot. Thus, each time a station transmits a packet, the unsuccessful transmission probability is constant at steady state in a generic slot. This is a valid assumption if backoff stage of the whole system with nodes and random bit errors is at steady state. The details of the assumptions and derivation of the model can be found in [14] and [15].

In terms of fragmentation overhead and random packet loss, suppose a MAC Service Data Unit (MSDU) of L bits is fragmented into j MAC Protocol Data Units (MPDUs), and denote L_{opt} for the length of the fragmented MPDUs. Subsequently, it incurs $j-1$ times the additional overhead of $H + 2\,SIFS + ACK$, where H is physical and MAC headers. Note that this overhead is additive, while the probability of packet loss is reduced exponentially from $1-(1-p_b)^L$ to $1-(1-p_b)^{L_{opt}}$, where p_b is bit error rate in the channel. This packet loss caused by wireless random errors has an immense impact on the probability of unsuccessful transmission, especially when the channel becomes worse. The probability of unsuccessful transmission increases as the channel becomes worse due to the influence of the random errors, which causes a transmission failure and exponential backoff, resulting in larger average waiting time. Therefore, the probability of collision is lower in this bad channel. However, since the probability of packet loss is greater in this channel, the overall probability of unsuccessful transmission is also greater, and the goodput decreases severely as the channel quality becomes worse.

From [14] and [15], we have goodput G, which is defined as the fraction of time that the medium is occupied to transmit user data successfully,

$$G = \frac{P_s \cdot (1 - P_e) \cdot T_s}{T_i + P_s \cdot (1 - P_e) \cdot T_f + (1 - P_s) \cdot T_c + P_s \cdot P_e \cdot T_f} . \qquad (1)$$

where P_s is the probability of successful transmission without collision and P_e is the probability of the packet loss by random bit errors in L bits packet. For the time components, T_i is the average idle time between two consecutive transmissions, T_c is the time needed to detect collision at the receivers, T_s is the time duration normalized to a slot time to transmit user data, and T_f is the time interval to send L bits successfully with fragmentation. Given a packet size L, the number of users n, and a BER, the solution, optimal fragment L_{opt}, of the nonlinear equation (1), can be uniquely determined using numerical approaches to find where G reaches the maximum value.

The optimal MPDUs for 1500 bytes MSDU are illustrated in Fig. 1 with various BERs and number of users for 1Mbps modulation rate in 802.11b. In a perfect channel with less than 10 users, fragmentation has no positive impact on goodput. Since the probability of random packet loss is negligibly small in this channel, the probability of unsuccessful transmission is almost equal to the probability of collisions. If the channel is perfect, the probability of collisions is not a function of packet length, but a function of number of users in the network. Therefore, if there is no hidden terminals and interference at the receivers, and all stations obey the basic access rules, a constant probability of collisions is expected regardless of the packet length, and the performance will be degraded gracefully. However, the probability of collisions increases as the number of user increases. As the contentions become severe, a longer packet needs more time to detect loss and recover from it. In such cases, the optimal fragmentation technique can provide more benefit by adjusting the fragment size to the channel.

As the channel becomes worse due to random packet drops, the fragment size should decrease abruptly to compensate for the random errors. The contention among users also affects the optimal MPDUs as mentioned earlier. However, the impact of the contention is minor, since random errors play a bigger role on the exponential backoff procedure than collisions caused by contentions. Consequently, optimal fragmentation improves goodput more effectively as the channel becomes worse, or the number of users increases in the network.

III. LINK ESTIMATION AND ADAPTATION

To apply optimal fragmentation dynamically in time varying channels, the sender should be informed of the SNR of the receiver. We use an unique on-demand adaptive SNR estimator to estimate the SNR of the receiver and use the value to compute the maximum rate and the optimal fragmentation in the equation (1). The network parameters considered in the model includes the incoming packet length, BER, number of users, and the transmission rates. In computing the optimal length to be used in the test-bed, MAC header and SNAP header should be considered. The problem is how to monitor these parameters in real time, such as incoming packet length, BER, number of users, and transmission rates, in order to decide the optimal fragmentation threshold and transmission rates without modifying the protocols. The packet length and the number of users are known parameters in the network. For the BER and transmission rates, a new rate adaptation algorithm is proposed to incorporate transmission rate with the optimal fragmentation to meet the desired BER or packet error rate (PER). In the following section, the rate adaptation is elaborated.

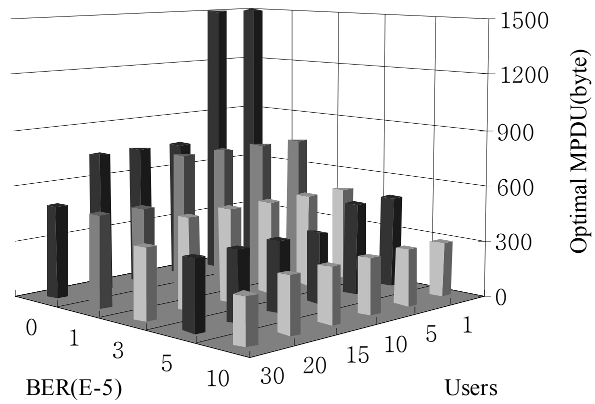

Fig. 1. Optimal MPDU for 1500 bytes MSDU with BER and number of users.

Since different modulation schemes support different rates, the rate can be adjusted to improve network goodput by switching to a higher modulation scheme if channel conditions improve. The BER vs. SNR curves can be found for 802.11 physical modes in [16], [17] and [18]. The target BER can be maintained by simply switching rates. With the consideration of the rate for a given SNR, the rate R may be written as

$$R = R_i, \; SNR_i \le SNR < SNR_{i+1}. \qquad (2)$$

where SNR_i is the minimum SNR to meet the target BER with the rate R_i. This is a conventional approach. However, the proposed rate switching algorithm is different in that the SNR range for the same transmission rate can be further shifted down to lower SNR range, if the optimal fragmentation improves packet drop rates such that the same goodput in the conventional rate switching can be obtained by using the optimal length MPDUs. That is

$$R_{opt} = R_{i\_opt}, \; SNR'_i \le SNR < SNR'_{i+1}. \tag{3}$$

where SNR'_i is the minimum SNR to achieve the same goodput for the conventional rate R_i. This rate switching can reduce the energy to be transmitted for the nodes, and increase the overall network goodput by reducing the interferences adjacent to the nodes.

A. Adaptive Channel Estimation

The challenging problem of SNR based rate switching and other approaches that use SNR of the receiver is to obtain the SNR, which is not supported in the 802.11 CSMA/CA MAC standard. In [2], [3] and [12], a modified RTS/CTS exchange is used to feed back the channel conditions of the receiver, which requires modifications of the protocol. The link adaptation strategies [8], [9] and [10] use the received signal strength (RSS) of the frame from the access point to select the best transmission rate for the sender. This approach assumes that the RSS has a linear relationship with the SNR of the receiver. However, this assumption is not valid when the AP supports multiple rates for downlink channels. Since mobile nodes may have different network cards, the transmission power of each user may be different. Therefore, SNR estimation with RSS for each station should be different, and the AP is not able to select proper rates individually for the stations. Furthermore, in the presence of interference at the receivers, strong RSS at the AP does not guarantee better SNR, and each user may experience different profile of interference.

The adaptive estimator in our algorithm uses on-demand, low overhead UDP messages to avoid modification of existing protocols. In the estimator, the received signal strength from the receiver is defined as $y(k)$. Suppose any mobile stations can overhear $y(k)$ as long as they are in the communication range. If the average received signal strength up to $k-1$ th frame is denoted as $\bar{y}_{RSS}(k-1)$, and the SNR estimation of the $k+1$ th frame is defined as $\hat{y}_{SNR}(k+1)$, the estimation of the SNR at the receiver can be represented as

$$\hat{y}_{SNR}(k+1) = (1-\gamma)\{\alpha\, \bar{y}_{RSS}(k-1) + (1-\alpha)\, y(k)\} + \gamma\, \bar{y}_{SNR}(k). \tag{4}$$

where $\bar{y}_{SNR}(k)$ is the average SNR of the receiver, and $0 \le \alpha \le 1$. The term, $\gamma \ge 0$, is to decide how much moving average of the received signal strength will be added to the average SNR of the receiver in the estimation. Given that the uplink and downlink channels are not always geographically symmetric, the estimation by only using the observed received signal strength is not valid for selecting L_{opt} and R_{opt}, even though there is no interference or hidden terminals at the receiver. However, the received signal strength is not totally irrelevant to the SNR of the receiver either. It provides a rough figure of the SNR in different time scale and amplitude in dBm. Thus, the adaptation algorithm should be informed of the initial average SNR of the receiver in any forms so that it tracks the SNR while reflecting the variation of the RSS on it. Since the signal power of RSS is in dBm, we use relative signal strength of the RSS in dB to match the units in (4) by subtracting a thermal noise level, which is typically –95 dBm.

In the estimator, the receiver informs the sender of $\bar{y}_{SNR}(k)$ using UDP messages in two events, i.e., when the difference between the averages of SNRs is greater than Δ_{SNR} (i.e., $|\bar{y}_{SNR}(k-1) - \bar{y}_{SNR}(k)| \ge \Delta_{SNR}$), or $\bar{y}_{SNR}(k)$ stays longer than the channel coherence time T_c [19], the time duration over which the channel impulse response is essentially invariant. That is

$$T_c = \sqrt{\frac{9}{16\pi f_m^{\,2}}} = \frac{0.423}{f_m}. \tag{5}$$

where f_m is the maximum Doppler shift. T_c may vary with respect to the BER performance of the modulation schemes and maximum Doppler shift. How quickly the estimator tracks the SNR can be determined by choosing the parameters α, γ, Δ_{SNR} and $\Delta_{coherence}$ in the equation (4). However, finding optimal parameters to tune the estimator for a typical wireless environment is a demanding task that requires experiments and field trials. We performed heuristic approach to select the parameters based on our experiments and observation.

In Fig. 2, the influence of UDP messages on the system throughput of the IEEE 802.11b mobile users is described. The vehicles transmit UDP control messages to estimate the SNR of the receiver during the TCP data transmission with 1500 bytes MSDU for 4% of packet error rate in NS-2 simulator. Even in the worst situation of 30 vehicles transmitting one UDP messages per second in average respectively, the performance loss that the on-demand adaptive estimator introduces is less than 1.6 % compared to the normalized throughput of the basic operation. However,

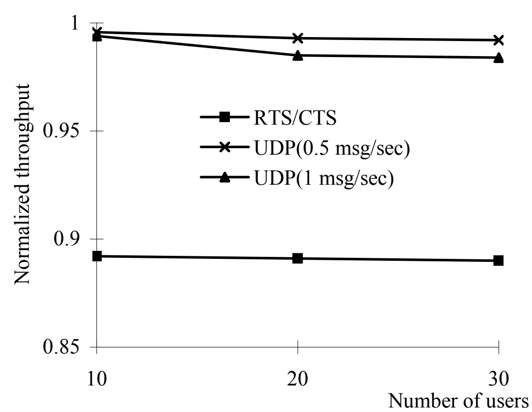

Fig. 2. Influence of overheads on throughput for the proposed On-demand adaptive estimator and RTS/CTS.

RTS/CTS based channel estimation incurs 11% of the throughput loss with the modifications of the protocols. Note that the proposed SNR estimation algorithm also can be incorporated with any link adaptation algorithms. Furthermore, the UDP message is relatively very short compared to data packet, and therefore it has better chance to survive in hostile environments. If the UDP message is lost, the sender simply maintains the previous SNR until next update is arrived successfully. Further overhead reduction of the estimator can be made by adjusting parameters in equation (4), if a coarse estimation is more desirable by sacrificing the accuracy in certain circumstances. In most cases, the average of less than one UDP messages per second can be obtained in vehicle-to-vehicle ad hoc communication (e.g., average of 0.71 messages per second). This overhead is almost negligible and has very little impact on the throughput. The details can be found in [21] with the results in real world experiments.

For the messaging overhead of UDP messages, even if the SNR varies severely in the channel, we have found that the estimation overhead produced by the on-demand UDP messages in the estimator yields less than a two-byte message in a second in general.

B. Other Considerations

The SNR of the receiver is one of the most important components in the calculation of L_{opt} and R_{opt}. Note that the target bit error rate or symbol error rate in SNR based rate switching are fixed. If the SNR fluctuates severely such that the BER also changes dramatically, the optimum rate adaptation automatically switches the transmission rate to a higher or lower rate in order to maintain the target BER or SER. Furthermore, selecting wrong optimal fragmentations are unlikely as the estimation algorithm has a very low approximation error (e.g., average of 0.373 dB for 6,000 samples in our experiment).

In the non-linear system given by (1), senders must determine L_{opt} and R_{opt}. Obtaining R_{opt} is straightforward if the senders can estimate the SNR precisely for a given target BER or SER. However, solving (1) to get L_{opt} in real-time is a computationally expensive task. By incorporating the knowledge that L_{opt} is a divisor of the original packet length, further simplification can be made to alleviate the complexity of the system for the real time implementation. Further reduction can be made to decrease the complexity of the system without performance degradation by setting a minimal packet length considered for fragmentation. For network security, it is the sender that computes L_{opt} and R_{opt}. Thus, if associated receivers are trusted entities, it would not introduce security problems. Additionally, senders could be any of the wireless stations, i.e., client stations and access points in WLAN, or wireless mobile stations in a multi-hop ad hoc networks and vehicle-to-vehicle networks.

IV. EXPERIMENT

Ad hoc is a network that requires no infrastructure to form connections to transmit data among nodes in the network. The stations within range can discover each other to form routing path by flooding or forwarding information to the other nodes. Thus, the network connections can be extended to multiple nodes to transport data through the routing path dynamically. If the nodes in the routing path move to the outside of the network, a new routing path is established to maintain connections using routing algorithms. These types of networks are very efficient especially in battlefields and other places where no central coordinators are available. However, it is a very fragile network compared to wired networks due to high mobility and the dynamic network environment. The proposed algorithm is a CSMA/CA based MAC layer technique, and can be applied to ad hoc networks that use the same MAC protocol.

The ad hoc experiment consists of four mobile stations equipped with 802.11b network cards. The network card is externally connected to the antenna. Each mobile station is running Ubuntu 6.10 [20] and MadWifi. All four mobile stations are configured to transmit in the same cell and channel to cause collisions in the network. The on-demand adaptive estimator is used to estimate the SNR of the receiver. The transmission power of the card is 5 dBm for the senders and the receivers.

For the performance evaluation, auto rate fallback [1] (ARF) protocol is used to compare the quality of service and goodput against the proposed algorithm. The reason for choosing ARF is that ARF is a default rate adaptation algorithm implemented in the 802.11 network card in the market. Unfortunately, more advanced rate adaptation algorithms such as [3] and [12] can hardly be implemented in the network cards due to the necessity of firmware changes, and therefore it is not feasible to compare these algorithms with the proposed algorithm in the experiments.

In the experiments, VLC media player 0.8.6f on mobile transmitters and receivers are installed for real-time streaming of mpeg1 video files encapsulated in MPEG Transport Stream (MPEG TS). VLC media player is a media player that provides multimedia player, encoder, and streamer supporting many audio and video codecs and file formats. Then, the recorded video frames at the receivers are compared to the original frames using video quality measurement tools. Both transmitters transmit 15 seconds of real-time video content, double wheel encapsulated in the RTP protocol in the office environment illustrated in Fig 3.

The transmitter of the proposed algorithm selects L_{opt} and R_{opt} dynamically using estimated SNR from the on-demand adaptive estimator, while ARF sender transmits normal 1364 bytes MSDU. For the estimator, α and γ are set to 0.9 respectively. The estimation parameter Δ_{SNR} and $\Delta_{coherence}$ are set to 0.2 dB. The SNR and the RSS are sampled in every 10 ms, and the average of 50 samples are

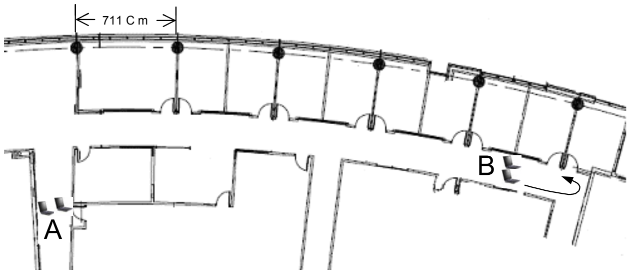

Fig. 3. Ad hoc experiment with four mobiles in office environments (Two transmitters and two receivers at 'A' and 'B')

used to calculate $\overline{y}_{RSS}(k-1)$ and 20 samples for $\overline{y}_{SNR}(k)$. We may use the BER equations found in [16] for an additive white Gaussian noise channel. The AWGN channel model is not realistic in ad hoc networks, but this model is useful for reference purposes. For DBPSK,

$$P_{DBPSK} = \frac{1}{2} e^{-E_b / N_0} \tag{6}$$

where E_b / N_0 is the SNR per bit. The approximated SER of DQPSK found in [17] is given by

$$SER \leq 2Q\left(\sqrt{\frac{E_s}{N_0}}\right) , \tag{7}$$

where E_s / N_0 is the SNR per symbol. The CCK is a variation of M-ary biorthogonal keying (MBOK) modulation for 5.5 and 11 Mbps, and the SER curve can be found in [18]. However, mobile environments where cars and trucks are driving with different curvature and elevation of the road are very different from the AWGN channel.

If there is a dominant line of sight (LOS) propagation path, Rician fading channel model may be applicable. When there is no dominant propagation along a LOS between the transmitter and receiver, the Rician factor K becomes smaller, and if the K equals to zero, it follows Rayleigh fading channel model, which is the worst wireless channel in mobile environments. From the theoretical BER vs. SNR analysis by [22], if the SNR is 35 dB, the BER is approximately 10^{-5} in Rayleigh fading channel for the IEEE 802.11b 1Mbps modulation. In a typical suburban outdoor mobile environment, the Rician factor K is approximately 10. If exact channel model is known, it is straightforward to use channel models to find boundaries between optimal fragmentation and transmission rates. Since underestimation of the SNR is more desirable to avoid using larger packets in a bad channel, Rayleigh fading channel model is assumed for the experiment. Based on the BER vs. SNR curve in [22], R_{opt} and L_{opt} can be determined using (1) and (3) such that the system goodput is always maximum over the entire range of the SNR. Note that R_{opt} is different from the conventional rate switching. For example, assume the target packet error rate is 8% as described in the standard for 1500 bytes MPDU. When the SNR of the channel improves, conventional rate switching algorithms change the rate from 1 Mbps to 2 Mbps at 18.87 dB to maintain the packet error rate below 8%, which corresponds to the SER of 1.4×10^{-5} in 2 Mbps. The maximum goodput of 1 Mbps is 0.833 Mbps up to the SNR of 18.87 dB. However, L_{opt} of 300 bytes at 2 Mbps yields 0.968 Mbps with the symbol error rate of 2×10^{-4} at 13.8 dB. Therefore, we can switch the rate to 2 Mbps at 13.8 dB to accomplish 0.968 Mbps, which yields 16.2% greater goodput than the maximum goodput, 0.833 Mbps, at 1 Mbps transmission rate. This rate switching technique provides benefits on the selection of a low interference routing path and efficient power management plan for limited battery capacity vehicles transmitting higher data rate.

In the experiment, two receivers at 'B' in Fig. 3 are moving away such that the SNR of the receivers are approximately 40dB. Then, they return to the senders in 'A'

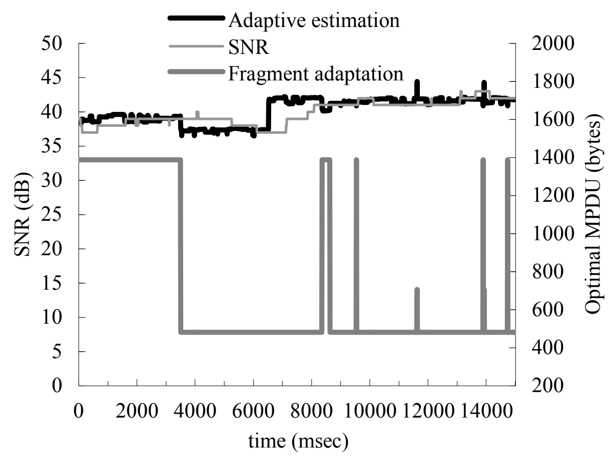

Fig. 4. Optimal adaptation of the MPDUs in the experiment.

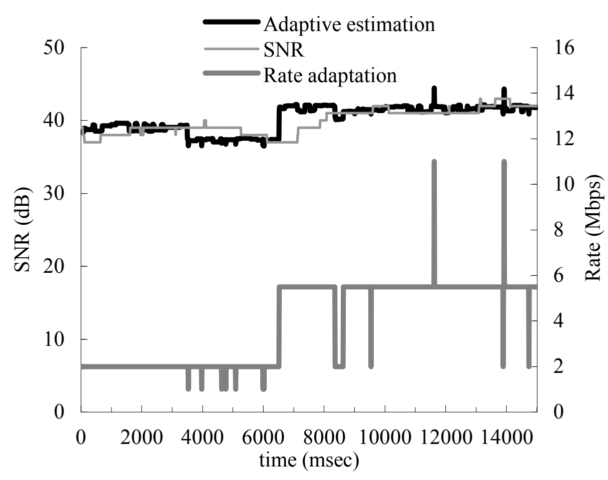

Fig. 5. Rate adaptation of the proposed algorithm in the experiment.

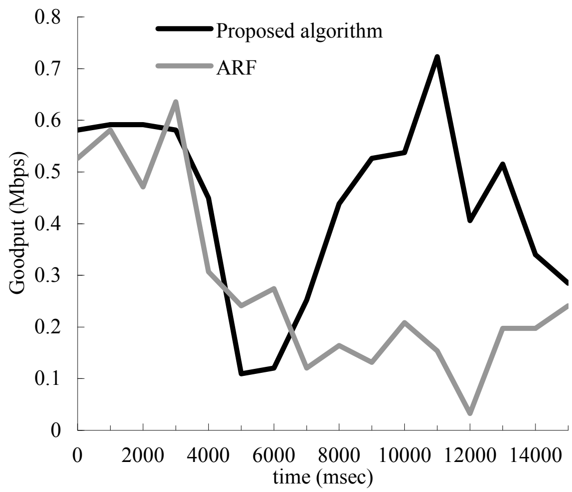

Fig. 6. Goodput of the proposed algorithm vs. ARF.

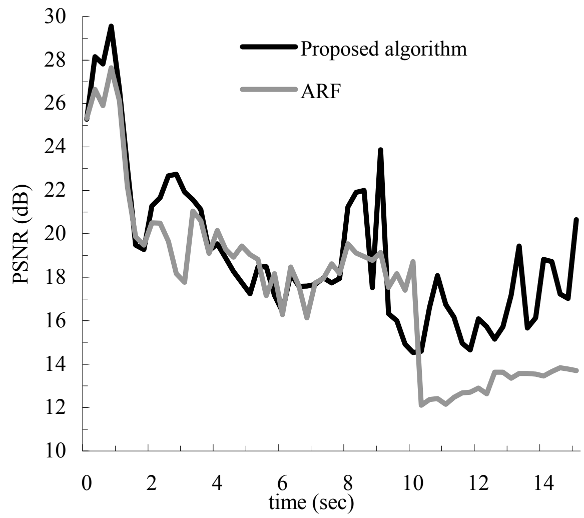

Fig. 7. PSNR comparison of the proposed algorithm vs. ARF.

in Fig. 3 at the end of the corridor. In the theoretical analysis from [16], [17] and [18], if the SNR is 40 dB, the channel is excellent to transmit and receive data without packet drops in an ideal AWGN channel.

Fig. 4, 5, and 6 illustrate the SNR of the receiver, the performance of the on-demand adaptive channel estimator, optimum rate and fragment adaptation, and the goodput of the proposed algorithm and ARF. The on-demand adaptive estimator tracks the SNR of the receiver with very low overhead (an average of 0.71 messages per second). The optimal MPDU L_{opt} follows the shape of the estimation in selecting MPUDs between 482 bytes, 706 bytes, and 1388 bytes. Since the actual SNR of the receiver is around 40 dB, the estimator models it as 20 dB, and the rate adaptation fluctuates between 1 Mbps to 11 Mbps in accordance with joint optimization of fragmentation adaptation. For 15 second in this experiment, the average goodput is 440.5 Kbps for the proposed algorithm while ARF is 280.2 Kbps as shown in Fig. 6, which yields 157.2% enhancement of the goodput. The results convince that the proposed algorithm is effective in noisy CSMA/CA wireless ad hoc networks.

In Fig. 7, the peak signal to noise ratio (PSNR) of the proposed algorithm and ARF is shown for the entire video sequences in RGB color spaces. PSNR is most commonly used measure to represent the difference of visual quality of two images. PSNR uses maximum pixel value of the images instead of signal power as in SNR. If there is no difference between two images, PSNR is infinite. The mean square error (MSE) to calculate PSNR can be represented as

$$MSE = \frac{1}{mn}\sum_{i=0}^{m-1}\sum_{j=0}^{n-1} \| I(i,j) - K(i,j) \|^2 \qquad (8)$$

for two m×n monochrome images I and K. Then, PSNR is defined as

$$PSNR = 20 \log_{10}\left(\frac{MAX_I}{\sqrt{MSE}}\right), \qquad (9)$$

where MAX_I is the maximum possible pixel value of the image. Normally if a PSNR is more than 30 dB, human eyes cannot distinguish the difference of the images. Typically, PSNR of 20dB or more is considered to be acceptable in wireless multimedia transmission [23]. In the Fig. 7, the proposed algorithm outperforms for almost entire duration of the experiment, and especially when the channel becomes worse, significant enhancement of quality of service can be achieved. The captured video images of the experiment at high and low PSNR channel are illustrated at 1 second and 15 seconds mark in Fig. 8 and 9. As we can see from these results, the proposed algorithm has more benefit as the

Fig. 8. The captured video images of high PSNR channels at 1 second in the ad hoc network experiment (from left: Original, Proposed and ARF)

Fig. 9. The captured video images of low PSNR channels at 15 second in the ad hoc network experiment (from left: Original, Proposed and ARF)

channel becomes worse, while maintaining the same quality of service in a high PSNR channels.

Although this experiment is performed for a single-hop ad hoc network with four mobile stations, it can be easily extended to a general multi-hop ad hoc networks and the margin of performance enhancement will be increased as the number of hops increases owing to the existence of possible packet drops and interferences along the routing path [14].

Further analysis of the proposed model can be made through combining other wireless network parameters, such as mean excess delay and RMS (Root Mean Square) delay spread to define multipath components in this ad hoc channel. Many vendors require a 65-ns RMS delay spread to support full data rate in 802.11b. However detailed modeling of the wireless channel associated with multipath components is beyond the research scope of this study, and can be conducted for the future work.

V. CONCLUSIONS

In this paper, an effective methodology to enhance the quality of real time multimedia streaming in CSMA/CA mobile ad hoc networks is presented. Using an adaptive on-demand SNR estimator, the algorithm dynamically selects the optimal fragmentation in time varying channels with minimal overhead. Through the experiments, we have found the system significantly enhances visual quality of the transmitted video and the goodput for approximately 157.2% in a single-hop ad hoc network. The proposed algorithms are applicable to next generation ad hoc communications, and realistic approaches that can be deployed without modification of existing standards.

REFERENCES

[1] Kamerman, A. and Monteban, L. "WaveLAN-II: A high-performance wireless LAN for the unlicensed band. *Bell Labs Technical Journal,* pp. 118-133, summer 1997.

[2] Holland, G., Vaidya, L., and Bahl, P. "A rate-adaptive MAC protocol for wireless networks," in *Proc. ACM MOBICOM'01,* Jul. 2001, pp. 236-251.

[3] Sadeghi, B., Kanodia, V., Sabharwal, A., and Knightly, E. "Opportunistic media access for multirate ad hoc networks," in *Proc. ACM MOBICOM'02,* Sep. 2002, pp.24-35.

[4] Lettieri, P. and Srivastava, M. B. "Adaptive Frame Length Control for Improving Wireless Network Link Throughput, Range and Energy Efficiency," in *Proc. IEEE INFOCOM '98,* March 1998.

[5] Ci, S. and Sharif, H. "Adaptive approaches to enhance throughput of IEEE 802.11 wireless LAN with bursty channel," in *Proc. IEEE LCN'00,* Nov. 2000, pp. 44-45.

[6] Tourrihes, J. "Dwell adaptive fragmentation: How to cope with short dwells required by multidedia wireless LANs," in *Proc. IEEE GLOBECOM'00,* VOL. 1, 2000, PP. 57-61.

[7] Tourrihes, J. "Fragmentation adaptive reduction: Coping with various interferers in radio unlicensed bands," in *Proc. IEEE ICC'01,* VOL. 1, 2001, PP. 239-244.

[8] Qiao, D. and Choi, S. "Goodput Enhancement of IEEE 802.11a Wireless LAN via Link Adaptation," in *Proc. IEEE ICC,* June, 2001.

[9] Qiao, D and Choi, S. and Shin, K. G. "Goodput analysis and link adaptation for IEEE 802.11a wireless LANs," in *IEEE/ACM Transactions on Mobile Computing,* vol. 1, pp. 278-291, Dec. 2002

[10] J. Prado and S. Choi, Link Adaptation Strategy for IEEE 802.11 WLAN via Received Signal Strength Measurement, in *Proc. IEEE ICC'03,* Anchorage, Alaska, USA, May 2003.

[11] Yin, J. Wang, X. and Agrawal, D. P., "Optimal Packet Size in Error-prone Channel for IEEE 802.11 Distributed Coordination Function," in *Proc. IEEE WCNC'04,* Atlanta, 21-25 March 2004.

[12] B. Kim, Y. Fang, T. Wong, and Y. Kwon, "Throughput Enhancement Through Dynamic Fragmentation in Wireless LANs," in *IEEE Transactions on Vehicular Technology,* Vol. 54, No. 4, July 2005.

[13] Bianchi, G. "Performance Analysis of the IEEE 802.11 Distributed Coordination Function," *IEEE Journal on Selected Areas on Communications,* Vol. 18, No. 3, March, 2000.

[14] Y.S. Chang, C. Lee and J.A. Copeland, "Goodput Optimization in CSMA/CA Wireless Networks," in *Proc. IEEE BROADNETS'07,* Raleigh, NC, Sep. 2007.

[15] Y.S. Chang, C. Lee, B.K. Kwon and J.A. Copeland, "Dynamic Optimal Fragmentation for Goodput Enhancement in WLANs," in *Proc. of the IEEE TridentCom '07,* Orlando, May. 2007.

[16] T.S. Rappaport, "Mobile radio propagation: Large-scale path loss," in *Wireless Communications: Principles and Practices.* Upper Saddle River, NJ: Prentice-Hall, 1996, pp. 69-185.

[17] J. G. Proakis and M. Salehi, "Differential phase modulation and demodulation," in C*omtemporary Communications Systems Using Matlab.* Piscataway, NJ: Brooks/Cole Thomson Learning, 2000, pp. 302-309.

[18] J. G. Proakis, "Performance of the optimum receiver for memoryless modulation," *Digital Communications, 3rd ed.*, New York, NY: McGraw-Hill, 1995, pp. 257-282.

[19] Rappaport, T. S., "Mobile radio propagation: Small Scale Fading and Multipath," in *Wireless Communications*. NJ: Prentice-Hall, 1996b, pp. 139-192.

[20] The Ubuntu Website (www.ubuntu.com), Accessed in May, 2007.

[21] Y. S. Chang, M. H. Lee, and John A. Copeland, "An Adaptive On-demand Channel Estimation for Vehicular Ad Hoc Networks," submitted to *IEEE CCNC,* 2008.

[22] P. Mahasukhon, M. Hempel and H. H. Chen, "BER Analysis of 802.11b Networks under Mobility," in *Proc. IEEE ICC*, 2007.

[23] Thomos, N., Boulgouris, N. V., and Strintzis, M. G., "Optimized Transmission of JPEG2000 Streams Over Wireless Channels," *IEEE Transactions on Image Processing*, vol. 15, No 1, pp. 54-67, January 2006.