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Improving goodput in IEEE 802.11 wireless LANs by using variable size and variable rate (VSVR) schemes

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Summary

In IEEE 802.11 wireless LANs, channel quality, network load, as well as the protocol itself are time-varying, limiting the goodput performance improvement in wireless LANs. Therefore, it becomes critical to dynamically adjust parameters of MAC and PHY layers according to variations of channel quality. In this paper, we propose variable frame size and variable data rate schemes for goodput enhancement. We first propose two optimal frame size predictors: a goodput regulator to maintain the committed goodput for non-greedy applications and an optimal frame size predictor for maximizing the goodput for greedy applications. Then, we propose a data rate drafting scheme and develop a variable size and variable rate (VSVR) scheme for further goodput improvement. Our extensive simulation results show that the proposed VSVR algorithm can double the channel goodput of current implementations. Moreover, the proposed scheme can be easily integrated with the current implementations of the wireless LAN MAC protocol. Copyright © 2004 John Wiley & Sons, Ltd.

KEY WORDS: wireless LAN; IEEE 802.11; link adaptation; goodput performance; Kalman filter

1. Introduction

With the fast-pacing deployment of IEEE 802.11 wireless LANs, there are more and more demands on supporting QoS in terms of throughput, delay and jitter in wireless LANs. When supporting QoS, a wireless LAN faces the same constraints as other wireless networks such as fading channel and power efficiency. On the other hand, IEEE 802.11 wireless LANs have their own constraints such as the contention-based CSMA/CA MAC protocol. In general, these constraints make the goodput performance more susceptible to transmission errors in wireless LANs than in other wireless networks such as cellular systems. When transmission error occurs, a retransmission is scheduled. Normally, retransmissions in wireless LANs are caused by collisions, bit errors or both. Collisions are caused by the nature of the contention-based MAC protocol and other problems such as hidden terminals, while bit errors are caused by time-varying channel quality due to the existence of interference, noise and fading. At the same time, most of the current communication systems use a set of fixed system parameters such as data rate and frame size, which is obviously not efficient in a dynamic environment.

In recent years, link adaptation has been proposed for improving system performance by adaptively changing the protocol parameters according to channel quality and network load. In References [1–3], the optimal frame size prediction has been studied. The basic idea is to get the maximum throughput by

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dynamically changing the frame size according to the variations of channel quality. When the channel quality is good, a larger frame size can be used to get a higher throughput performance. A shorter frame size is otherwise adopted to lower the number of retransmissions. In Reference [4], link adaptation has been addressed from several aspects such as frame size, equalizer design and power control in a Rayleigh fading environment. Recently, fragmentation threshold adaptation algorithm has been proposed and discussed. When interferences appear, a long frame could be divided into several short fragments at the transmitter side and they will be reassembled at the receiver side after all fragments are correctly received. In Reference [5], optimal contention window size algorithms were proposed and discussed. It showed that the throughput performance could be changed significantly by optimizing protocol parameters. Adaptive modulation schemes over fading channels have been studied extensively [6–9]. In general, adaptive modulation focuses on improving spectral efficiency and channel throughput with desired bit error rate without losing power efficiency at the same time. Among these research works, some of them are specially based on wireless LANs. However, none of them has fully considered the source (application) characteristics in their schemes for throughput enhancement in wireless LANs. It is clear that for different applications, goals for using optimal frame size may be different. For instance, for greedy applications such as ftp, one wants to transfer the file as fast as possible. Therefore, the goal for using optimal frame size is to achieve the highest possible throughput, while for non-greedy type of applications such as voice and video, the goal for optimizing frame size is to maintain the committed throughput. So we can conclude that different applications may need different types of optimal frame size predictors.

In this paper, we propose variable frame size and variable data rate schemes for goodput enhancement. We first propose two optimal frame size predictors: a goodput regulator to maintain the committed goodput for non-greedy applications and an optimal frame size predictor for maximizing the goodput for greedy applications. Then, we propose a data rate drafting scheme and develop a variable frame size and variable data rate scheme for further goodput improvement. Extensive simulations have been carried out and results show that our proposed schemes can improve the goodput performance by 50% more than the scheme using fixed frame size and data rate. Furthermore, we can observe that the variable frame size and variable data rate scheme can improve the goodput performance by 25% more than the proposed frame size predictors.

The rest of this paper is organized as follows. A review of throughput performance of IEEE 802.11 protocol is given in Section 2. The proposed frame size predictors for greedy and non-greedy sources are presented in Sections 3 and 4 respectively. Simulation settings and results are given and analyzed in Section 5. We have some concluding remarks in Section 6.

2. Brief of the IEEE 802.11 Wireless LAN

The IEEE 802.11 standard focuses on the issues of MAC and PHY layers, providing a current style of IEEE 802 LAN over wireless medium. Figure 1 is an illustration of current IEEE 802.11 WLAN. According to the standard [10], two basic operation modes have been specified, which are infrastructure mode and ad hoc mode. Infrastructure mode uses basic service set (BSS), where all communications occur via an access point (AP); ad hoc mode uses independent basic service set (IBSS), where stations can communicate with each other directly without using APs.

Moreover, there are two channel access methods defined in the current 802.11 protocol, the distribution coordination function (DCF) and the point coordination function (PCF). DCF is actually CSMA/CA contention-based method and PCF is a polling-based contention-free channel access method. Due to its complexity and lack of clear specification of PCF, there are few PCF implementations nowadays. Thus, in this paper, we focus on the DCF for our study.

Figure 2 illustrates the timing of DCF. Due to the difficulties of collision detection in wireless channel, CSMA/CA channel access method is adopted in DCF, which combines physical carrier sensing and virtual carrier sensing together to avoid collisions. Every station that wants to transmit packets will sense the channel first, if both physical carrier sensing and virtual carrier sensing indicates that channel is idle for a period of DCF inter-frame spacing (DIFS), the transmission will start. Otherwise, it will experience a random backoff time. When a station receives a frame correctly, after a short inter-frame spacing (SIFS), and if necessary an acknowledgement (ACK) is sent back to the sender. In the 802.11 protocol, there is no explicit notification if frame errors happened. If a scheduled acknowledgement is not received the sender will assume errors happened and a retransmission,

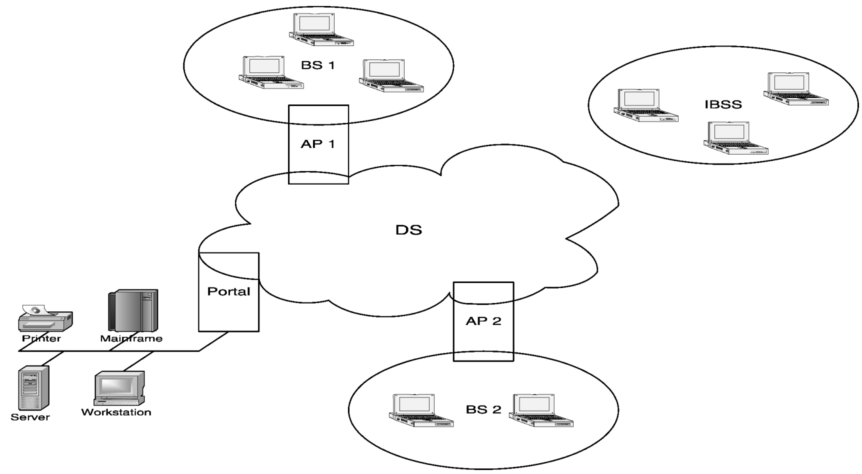

Fig. 1. Typical IEEE 802.11 WLAN topology.

if required, will be arranged. In wireless LAN, retransmissions may be caused by frame errors and collisions. Overall, at the time of this writing, there is still no QoS consideration and specification standardized for the current 802.11 standard.

Before going any further, we review the major difficulties of QoS provisioning in wireless LANs.

- Not like its wired counterpart, wireless channel is error prone and time varying due to the existence of slow fading, fast fading, path loss, shadowing, noises and interferences. This causes high frame error rate at receiver and results in retransmissions. Consequently, the channel efficiency is severely degraded.
- Bandwidth of wireless networks is always scarce and valuable. IEEE 802.11b WLANs only have three totally separated channels. To achieve higher throughput from limited bandwidth, advanced modulation schemes like CCK modulation scheme have been adopted. But normally these advanced modulation schemes require higher signal-to-noise-ratio (SNR) at receiver. Even in a LOS environment, the SNR performance can be exaggerated by many factors such as interferences caused by different operation modes (IBSS and BSS), network planning problems, co-channel interference and overlapping BSSs.
- Throughput is an important QoS parameter to evaluate the performance of wireless network. Throughput is vulnerable to variations in channel quality, packet length, lower layer protocol efficiency, network load and so forth. Goodput, defined as the ratio of achieved user data rate over channel raw data rate, is normally employed to give more accurate throughput evaluation. The overhead in IEEE 802.11 mainly comes from back off, deference, MAC and PHY layer header, management frames and control frames and retransmissions.

From above discussions, it is clear that goodput is very important for QoS provisioning in wireless LANs. Naturally, it leads to another important question—how to improve goodput in IEEE 802.11 wireless LANs. In the following sections, we will propose two frame size predictors for a higher throughput performance. We will also pursue further performance enhancement by integrating them with a data rate drafting algorithm.

3. Optimizing Frame Size by Maintaining a Committed Maximum Goodput

3.1. Brief of Kalman Filter Fundamentals

Considering a general prediction problem with noisy data,

$$x_{k+1} = F[x_k, u_{k+1}, w_{k+1}] \tag{1}$$

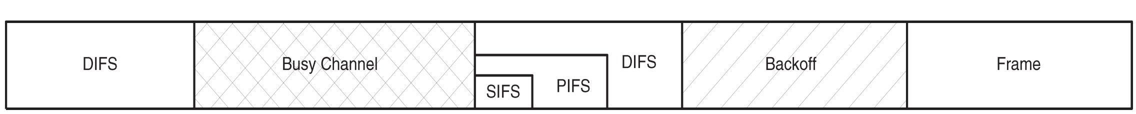

Fig. 2. Basic access method.

In this equation, x_k is the system state at the time k. u_{k+1} is the system control input at the time $k+1$. w_{k+1} is the process noise at the time $k+1$, which is assumed to be additive process noise. $F(.)$ is the process model. For the observation system, the following equation is used.

$$z_{k+1} = H[x_{k+1}, u_{k+1}, v_{k+1}] \tag{2}$$

Here, $H(.)$ is the observation model, $v_{k+1}$is assumed to be additive observation noise. V and W could be any kind of distribution, but generally they are uncorrelated. Our main concern is to estimate system states X by using noisy observations Z under the known process model and observation model.

When $F(.)$ and $H(.)$ are linear systems, the Kalman Filter can be used to provide prediction with least mean squared error (LMSE) of true system states recursively. The Kalman filter is widely adopted for different usages such as prediction, estimation and smoothing in many fields. The advantage of the Kalman filter is that it is an efficient computational recursive solution with the LMSE [11,12].

The Kalman filter uses a predictor–corrector structure. The predictor predicts the next system state through the processing model. The corrector will update the Kalman gain and will then observe the new measurement through the observation model. A posteriori prediction of the system state can be derived from the Kalman gain, *a priori* state and the measurement of the updated system state. The Kalman filter can be represented by the following set of equations.

The processing model is

$$x_{k+1} = Ax_k + Bu_{k+1} + w_{k+1} \tag{3}$$

and the observation model is

$$z_{k+1} = Hx_{k+1} + v_{k+1} \tag{4}$$

The Kalman gain is

$$K_{k+1} = P_{k+1}H_{k+1}^T(H_{k+1}P_{k+1}H_{k+1}^T + R_{k+1})^{-1} \tag{5}$$

Here, P_{k+1} is *a priori* prediction error covariance that can be written as

$$P_{k+1} = E[(x_{k+1} - x_{\hat{k}+1})(x_{k+1} - x_{\hat{k}+1})^T] \tag{6}$$

and the a posteriori update of x_{k+1} is

$$x_{\hat{k}+1} = x_{k+1} + K_{k+1}(z_{k+1} - H_{k+1}xk + 1) \tag{7}$$

3.2. Optimal Frame Size Predictor Based on the Kalman Filter

In wireless LANs, damaged frames are discarded and retransmitted. As a result, the overall system performance is degraded greatly. The performance of wireless system is very sensitive to the frame size. The larger the frame size, the higher the frame error rate is.

The channel efficiency has the following relation with frame size [13].

$$\gamma = \frac{1}{(1 - P_\mathrm{b})^{-L}} \tag{8}$$

where γ is the channel efficiency, $L = l + h$ is the total size of a frame, l is the payload size of a frame, h is the header size of a frame and P_b is the bit error rate under a certain channel quality. Since in the wireless channel errors often occur in bursts, we assume that equalizer is employed in wireless systems. Thus, bit errors occur independently.

Figures 3 and 4 show relations among bit error rate, frame length and backoff time in wireless LANs. According to Reference [1], there is an optimal frame size, given a certain channel quality and network load. In the IEEE 802.11 wireless LAN, the optimal frame size is greatly affected by the backoff time, the bit error rate and the network load. Thus, it is difficult to track the optimal frame size in the wireless LAN.

In order to find accurate optimal frame sizes, we adopt the Kalman filter to predict the optimal frame size. The following equations are developed as the state model of the proposed frame size predictor.

$$\gamma_k = \frac{1}{(1 - P_{\mathrm{b}k})^{-L_k}} \tag{9}$$

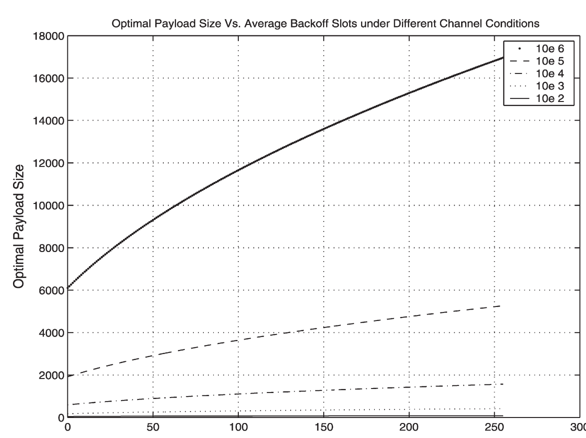

Fig. 3. Optimal payload size versus average backoff slots.

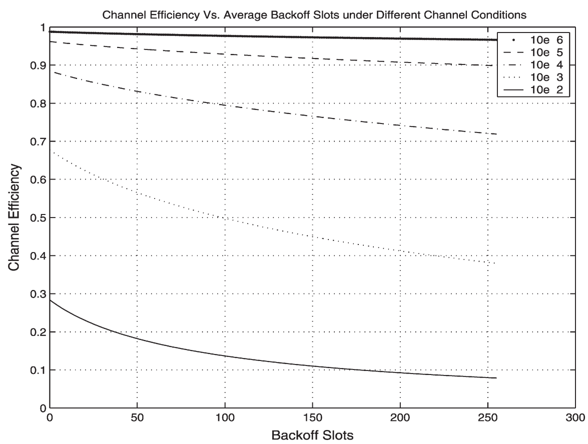

Fig. 4. Channel utilization versus average backoff slots.

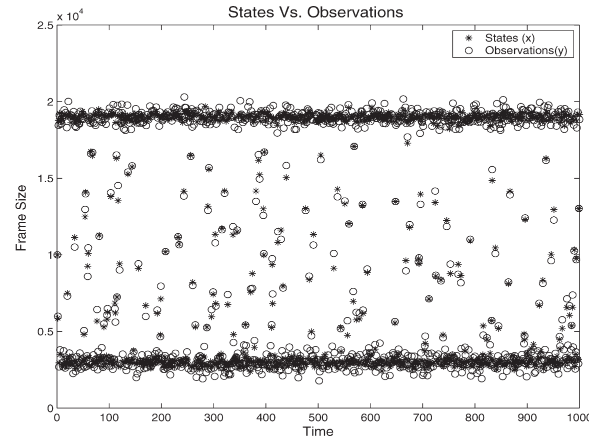

Fig. 5. State versus observation.

At the time $k+1$, we have

$$\gamma_{k+1} = \frac{1}{(1 - P_{\mathrm{b}k+1})^{-L_{k+1}}} \tag{10}$$

Note that our goal here is to use the optimal frame size to maintain the desired channel efficiency γ_k. Hence, the following equation can be derived by combining the above two equations.

$$L_{k+1} = L_k \frac{\log(1 - P_{\mathrm{b}k})}{\log(1 - P_{\mathrm{b}k+1})} \tag{11}$$

Normally, $P_{\mathrm{b}}k$ is much smaller than, i.e., $P_{\mathrm{b}}k \ll 1$, using the approximation $\log(1 + P_{\mathrm{b}k}) \simeq P_{\mathrm{b}k}$, the above equation can be written as

$$L_{k+1} = L_k \frac{P_{\mathrm{b}k}}{P_{\mathrm{b}k+1}} \tag{12}$$

Now we get the frame size prediction model. In general, most communication network systems have restriction on frame sizes such as $L_{\max}$ and $L_{\min}$. Thus, The the state model of the frame size predictor can be expressed as following equations.

$$L_{k+1} = \begin{cases} L_{\max} & L_{k+1} > L_{\max} \\ L_k \frac{P_{\mathrm{b}k}}{P_{\mathrm{b}k+1}} & L_{\min} < L_{L+1} < L_{\max} \\ L_{\min} & L_{k+1} < L_{\min} \end{cases}$$

For the observation model, we choose

$$Z_{k+1} = Q(L_{k+1}) \tag{13}$$

Z_{k+1} is the observation at time $k+1$ and $Q(.)$ is the observation quantization function. The observation model depends on system processing limitations and considerations of simple and efficient implementations. In this case, the processing noise and observation noise is mainly caused by the accuracy of system modeling. Since our main goal here is to evaluate the effectiveness of the proposed approaches, we use the off-line method to determine these values. In this paper, the variances of processing and observation noises are set to 400 bits. For future operational tuning, more measurements should be pursued.

The total samples in the simulation are 10^5. The results of the proposed algorithm are compared with that of the moving average method in Figures 5 and 6. Table I shows the RMS error of different prediction schemes, compared with the minimum RMS error of the fixed frame size scheme. In this simulation, we chose frame sizes from 1150 to 18 400 bits with an increase step of 10 bits. The frame size that achieved the minimum RMS error is listed in the table.

Figure 6 shows the comparison of different prediction methods. From simulation results, we can

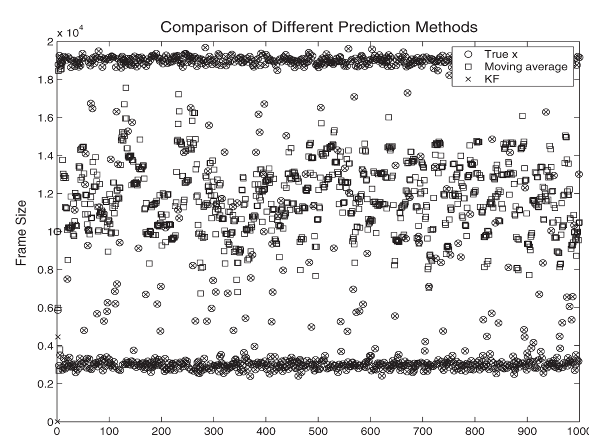

Fig. 6. Proposed predictor versus moving average (window = 10).

Table I. Comparison of different prediction methods.

Prediction method	RMS error
Proposed predictor	106.05
Moving average (window = 10)	3437.9
Fixed frame size scheme	7138.3

conclude that the RMS error of state samples predicted by using the proposed predictor is lower by an order of magnitude than that of the moving average method.

However, the above results are derived under the assumption that perfect channel estimations are available, which does not exist in practice. In order to evaluate our proposed frame predictor with imperfect channel estimations, we run the simulations by introducing channel estimation errors. Therefore, the process model can be described as

$$L_{k+1} = L_k \frac{\hat{P}_{\mathrm{b}k}}{\hat{P}_{\mathrm{b}k+1}} \tag{14}$$

where $\hat{P}_{\mathrm{b}k}$ is the noisy channel estimate at time k.

In the simulations, the range of interest is between 10^{-2} and 10^{-5} and the estimation of a probability of bit error is derived from 10^6 samples. Table II shows the mean and the 95% confidence intervals of the estimated bit error rate.

We run the simulation with a random estimation error in bit error rate. We then get the results listed in Table III. From Table III, we can observe that the performance of the proposed predictor is degraded due to the estimation error of bit error rate, but it can still provide a better performance than the moving average method.

Table II. Mean and 95% confidence intervals of channel estimation.

BER	Mean	95% confidence intervals
10^{-5}	1.1593×10^{-5}	(4.7954×10^{-6}), 1.8390×10^{-5}
10^{-2}	1.0001×10^{-2}	(9.8059×10^{-2}), 1.0197×10^{-2}

Table III. Comparison of RMS error with channel estimation error.

Prediction method	RMS error
Proposed predictor	1784.7
Moving average (window = 5)	3774.3
Moving average (window = 10)	4174.4
Moving average (window = 100)	5065.2
Fixed frame size scheme	7138.3

4. Optimizing Frame Size by Maximizing Goodput

Goodput is one of the most important parameters to evaluate the performance of wireless LANs and it is defined as the value of data payload over the duration between two renewal points [14]. Renewal points only occur when a transmission has been acknowledged successfully and the successive deference procedure is finished.

$$\rho = \frac{L}{\tau} \tag{15}$$

where ρ is the goodput of each station in wireless LAN system, L is the payload of a frame and τ is the duration between two renewal points. In general, the length of renewal interval is dependent on the number of retransmissions and the network load of the whole service area. Retransmissions may be caused by collisions and frame errors; the network load could be affected by the number of users and the type of applications. In this paper, we assume that each station associated with the same BSS always has packets to be sent, i.e. we only consider the saturated performance. We also assume that the channel quality is not changed during the period of a transmission.

According to the IEEE 802.11 MAC protocol, the renewal interval $(\tau_{\max})$ between two renewal points is composed with two time durations: the first part comes from retransmissions caused by collisions (τ_{coll}) and the second part comes from retransmissions caused by bit errors (τ_{ber}). So we have

$$\tau_{\max} = \tau_{\mathrm{coll}} + \tau_{\mathrm{ber}} \tag{16}$$

When a collision happens, the sender will proceed with a DIFS and a random backoff time. Moreover, collisions will make other stations inside a BSS defer a period of time that is equal to at most the time duration used to send a frame. In an error-free channel, the number of collisions only depends on the average backoff time slots [5], which is decided by network load, i.e. the number of stations associating with the same BSS. In the IEEE 802.11 protocol, a backoff procedure is invoked if an ACK times out, the PHY medium is still busy after the current backoff time has elapsed or a frame error occurs. In either of these cases, the minimum contention window size will be increased exponentially, capped by the maximum value [10]. So τ_{coll} can be written as

$$\tau_{\text{coll}} = \frac{(L + H + \text{Coll})N}{R} + (B + D + h + o)N \quad (17)$$

In the above equation, L is the payload size of a frame and H is the MAC protocol header of a frame, h is the preamble overhead introduced by PHY layer, o is the protocol overhead such as MAC and PHY processing delay and Tx/Rx switching time, R is the transmission data rate and B is the average number of random backoff time slots under a certain network load and channel quality. The backoff procedure follows the algorithm specified by the IEEE 802.11 wireless LAN standard. N is the average number of collisions occurred between two renewal points. *Coll* is the average length of collisions, D is DIFS whose values are specified by the standard. Similarly we can express τ_{ber} as following

$$\begin{aligned}\tau_{\text{ber}} = &\frac{(L + H)(1 - P_{\text{b}})^{(L+H)}}{R} + (B + D + h + o)\\ &\times (1 - P_{\text{b}})^{(L+H)} + \frac{\text{ACK}(1 - P_{\text{b}})^{\text{ACK}}}{R}\\ &+ (S + h + o)(1 - P_{\text{b}})^{\text{ACK}} \end{aligned} \quad (18)$$

In this equation, P_{b} is the bit error rate under a certain channel quality. ACK is the frame length of ACK frame. S is the duration of SIFS. Here we write the average retransmissions caused by bit errors as $(1 - P_{\text{b}})^{FL}$ and FL is the length of frame. So we get

$$\begin{aligned}\tau = &\frac{(L + H)((1 - P_{\text{b}})^{(L+H)} + N)}{R} + (B + D + h + o)\\ &\times ((1 - P_{\text{b}})^{(L+H)} + N) + \frac{\text{ACK}(1 - P_{\text{b}})^{\text{ACK}}}{R}\\ &+ (S + h + o)(1 - P_{\text{b}})^{\text{ACK}} + \frac{N \times \text{Coll}}{R} \end{aligned} \quad (19)$$

We use the largest frame size as the duration of collision in the above equation. Therefore, the above equation turns into the upper bound of the renewal interval. Consequently, the lower bound of goodput can be expressed as

$$\begin{aligned}\rho = &LR/((L + H)((1 - P_{\text{b}})^{(L+H)} + N)\\ &+ (B + D + h + o)((1 - P_{\text{b}})^{(L+H)} + N)R\\ &+ \text{ACK}(1 - P_{\text{b}})^{\text{ACK}} + (S + h + o)\\ &\times (1 - P_{\text{b}})^{\text{ACK}}R + N \times \text{Coll}) \end{aligned} \quad (20)$$

Before going further, we analyze the above equation. From the equation, we conclude that in wireless LANs, the higher the data rate used, the lower is the channel utilization achieved. This is true since using higher data rates, the equivalent overhead, consisting of backoff time slots, DIFS and SIFS, is getting larger. However, for the end-to-end delay, using the highest possible data rate to transmit data can provide the best throughput performance when the channel is clear, our simulation results in Section 4 prove this. On the other hand, lower data rates can achieve higher coding gain than higher data rates. For example, 11 bit Barker code is used in 1 and 2 Mbps data rate schemes, which greatly improves the reliability of transmissions. Thus, lower data rates are preferable when the channel quality is bad.

In summary, adaptive algorithms have to be adopted for goodput enhancement in wireless LANs, since both the number of wireless stations inside a BSS and the channel quality are time varying, the goodput is changing as a function of the number of users associated with the same BSS(M) and the channel quality (P_{b}) and the frame payload size (L).

Even though adaptive frame size algorithms have been investigated on the context of different kinds of wireless networks [1,2], but none of them considers the question of how optimal frame size is affected by both the number of stations and the channel quality in the context of IEEE 802.11 wireless LANs. In this section, we calculate the optimal frame size by maximizing the Equation 20.

$$\frac{\partial\rho(L, Pb, M)}{\partial L} = 0 \rightarrow L_{\text{opt}} \quad (21)$$

In practice, most communication systems have restrictions on frame sizes by specifying $L_{\max}$ and $L_{\min}$. Here, the following equations are developed as the frame size predictor.

$$L_{\text{opt}} = \begin{cases} L_{\max} & L_{\text{opt}} > L_{\max} \\ L_{\text{opt}} & L_{\min} < L_{\text{opt}} < L_{\max} \\ L_{\min} & L_{\text{opt}} < L_{\min} \end{cases}$$

Note that the L_{opt} is a local optimum since the channel environment is changing in terms of bit error rate and the number of users. We claim that this optimality keeps valid at least between two renewal points. This claim is supported by the channel measurements of indoor wireless channels [15], where the channel quality could be kept at the same level during at least several milliseconds if an equalizer is adopted at the receiver. As a result, we can use L_{opt} as the optimal frame size for the next data frame transmissions. Because of the CSMA/TDD nature of the IEEE 802.11 wireless LAN, it is unnecessary to use the feedback channel to exchange information used in predictions such as channel qualities at receivers. It is also very easy for the receiver to know how many users are associated with a BSS from its access control list (ACL).

5. Data Rate Drafting Algorithm for Further Goodput Enhancement

In most of the today's Internet applications, we only care about the throughput and delay performance of a given system. The goal is to send or receive a file as fast as possible. So according to previous analysis using the highest data rate is the effective way in most current IEEE 802.11 system implementations. Therefore, most current wireless LANs are designed to achieve a desired SNR for the highest data rate at receiver, meaning that the receiver sensitivity is fixed. But the wireless channel quality is highly dynamic due to mobility, antenna direction and interferences from other systems like microwave ovens, cordless phones and co-existence of IEEE 802.11 system and Bluetooth [16], causing the receiver sensitivity cannot be guaranteed now and then.

There are three different ways to keep the designed receiver sensitivity stable when the channel quality is bad. The first solution is to increase the transmission power. The second is to lower the data rate because the lower data rate has more coding gain. The third is to use the fragmentation or use smaller frame size. It is clear that increasing the transmission power is not the solution since this causes the interference to other BSSs and consumes more power which is unacceptable for portable or mobile devices such as pocket PCs or laptops. Thus, we propose the following data rate drafting algorithm to increase the goodput and delay performance even when wireless channel is in deep fades. Through this algorithm we try to keep the use of the highest possible data rate under a certain channel and network load to maximize the channel throughput.

Variable Data Rate and Variable Frame Size Algorithm:

Input: Data rate (R_k) supported by IEEE 802.11 wireless LAN
$R_k \in [1, 2, 5.5, 11]$.
$L_{\min} \leq L \leq L_{\max}$ the frame payload size, and
$L_{\min} \leq M \leq L_{\max}$ the number of users associated with the same BSS.
ρ the current goodput.

Output: Frame payload size and Transmission Data Rate used in the next transmission.
Local Vars: the k is an integer variable that denotes the kind of data rate.
Initialize $k := 4$

if ACK is time out or $\rho \leq R_k$
k = k − 1 and choose R_k as the data rate used in the next transmission;
compute L_{opt} by Equation 21;
transmit a frame with payload length of L_{opt} using data rate R_k;

else
k = k + 1 and choose R_k as the data rate used in the next transmission;
compute L_{opt} by Equation 21;
transmit a frame with payload length of L_{opt} using data rate R_k;

end-if

6. Results and Analysis

In this section, we will present the simulation results and analyze three algorithms: the variable frame size and variable data rate algorithm (VSVR), the fixed frame size and variable data rate algorithm (FSVR) and the variable frame size and fixed data rate algorithm (VSFR).

6.1. Simulation Settings

The simulation parameters are chosen according to the IEEE 802.11b wireless LAN specification (1999) [10]. Table IV gives the values of protocol parameters specified in the current IEEE 802.11 standard.

In our simulations, the channel quality samples are taken from the field measurements of a typical indoor office [15], illustrated in Figure 7. The receiver sensitivity is −80 dBm, which satisfies the requirement that it needs to be less than 8% FER at the receiver with 1024 byte frame size [10]. There are two channel quality scenarios used in this simulation, the noise floor −89 dBm is chosen as a bad channel scenario and the noise floor −95 dBm is chosen as a good channel scenario.

The average goodput ρ is measured by evaluating how fast to transfer 1MB file. The following

Table IV. MAC and PHY Parameters for the IEEE 802.11 Standard.

SIFS	10 μs
DIFS	50 μs
Time slot	20 μs
Data rate	1 Mbps, 2 Mbps, 5.5 Mbps, 11 Mbps
MAC header length	214 bits
ACK length	112 bits
Maximum Frame Length	2304 bytes
Preamble length	192 μs(long) and 96 μs(short)
MAC delay	≤ 5 μs
PHY delay	≤ 10 μs
CWmin	32
CWmax	256

Equation 22 is used for calculating the performance parameters.

$$\begin{cases} \rho = \frac{\iota}{\tau} \\ \tau = \sum_{i=1}^{N} \omega_i \\ \omega_i = \mu + DIFS + B_i + \dfrac{\nu + k_i + \text{ACK}}{R_i} \end{cases} \tag{22}$$

In the above Equations, ι is the total payload bits need to be transferred, equal to the file size; N is the total transmission number; R is the transmission data rate. τ is the total transmission time. ω_ι is the total transmission time needed by successfully transmitting a frame. μ is the PLCP overhead and ν is the MAC overhead. κ is the payload size of each frame. B_ι is the backoff time experienced by a frame. ACK is the length of an ACK frame in bits. R_ι is the data rate used by the ith transmission.

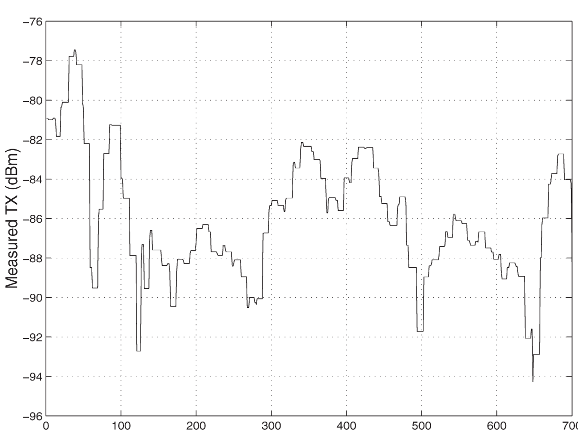

Fig. 7. Channel measurements in a typical indoor environment.

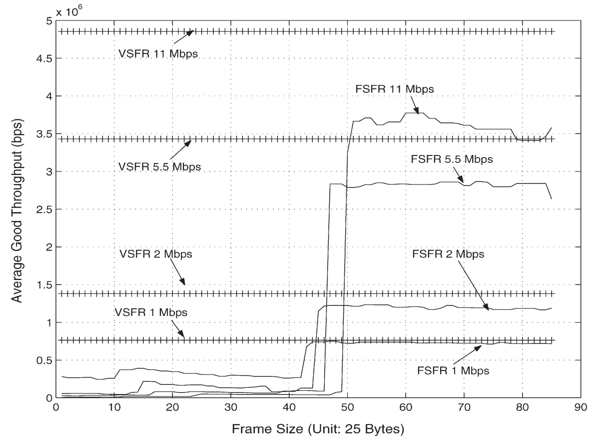

Fig. 8. Goodput performance of VSFR with five STAs and NF = −95 dB.

6.2. Algorithm Performance

Figure 10 shows that the goodput performance of the VSFR algorithm is compared with the throughput performance achieved by the current FSFR scheme. The noise floor is −89 dBm and there are five stations inside the service area. We can get the similar results with different network scenarios shown in Figures 8, 9, 10 and 11.

In all network scenarios, we observe that the proposed optimal frame size algorithm can greatly improve the goodput performance. Similarly, Figure 8 shows the average goodput of the proposed algorithm. Here, the noise floor is −95 dBm and there are five stations inside the same service area.

In the IEEE 802.11 wireless LAN, there exists the 'brick-wall' effect, meaning that we have to choose a frame size larger than a certain fixed frame

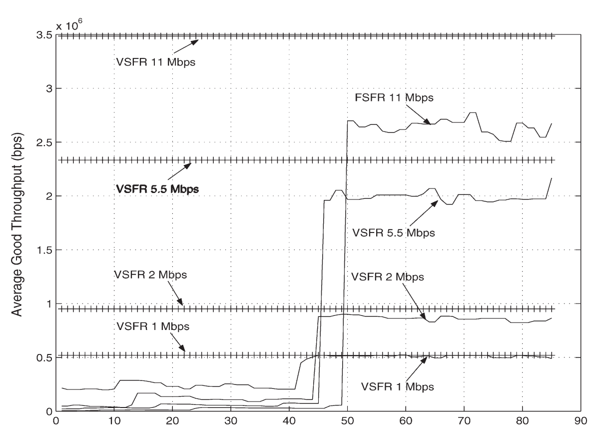

Fig. 9. Goodput performance of VSFR with 20 STAs and NF = −95 dB.

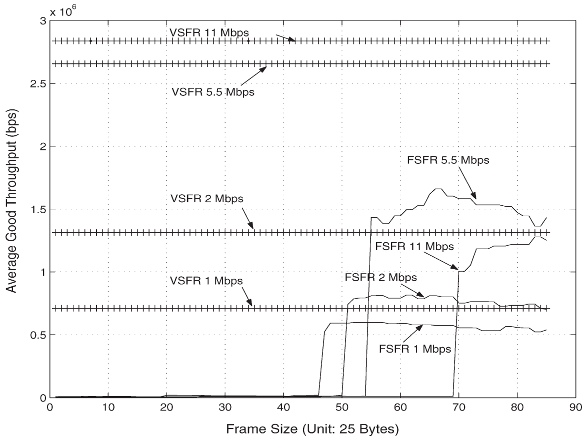

Fig. 10. Goodput performance of VSFR with five STAs and NF = −89 dB.

size threshold to offset the overhead and get a higher average goodput. In fixed data rate schemes, the higher the data rate, the more obvious the 'brick-wall' effect is. Moreover, we can also get the total overhead of each network scenario from reading the values of abscissa.

We observe that the proposed algorithms can improve the goodput performance and can eliminate the 'brick-wall' effect from appearing in fixed frame size algorithms.

The average goodput performance by adopting the FSVR algorithm, compared with the throughput performance of the conventional FSFR scheme, are illustrated in Figures 12, 13, 14 and 15. In this case, we use the variable-data-rate algorithm but keeping the frame size fixed. By using the FSVR algorithm, we

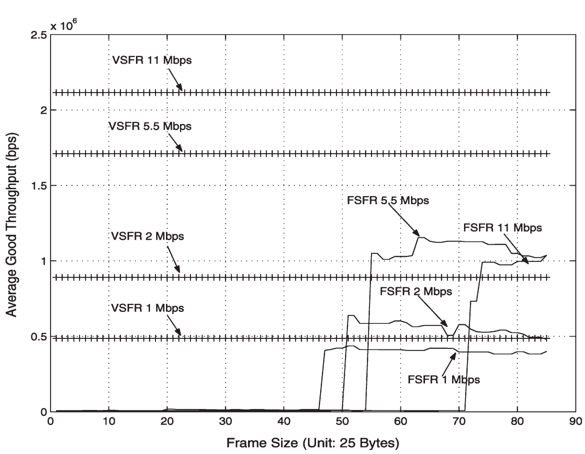

Fig. 11. Goodput performance of VSFR with 20 STAs and NF = −89 dB.

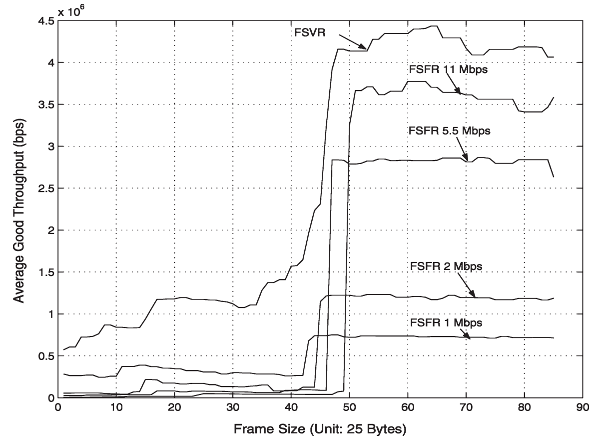

Fig. 12. Goodput performance of FSVR with five STAs and NF = −95 dB.

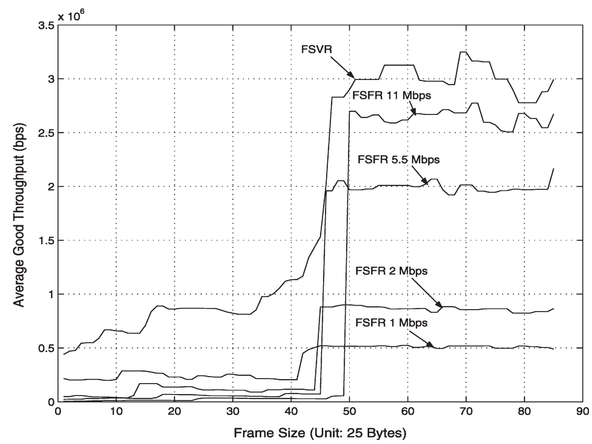

Fig. 13. Goodput performance of FSVR with 20 STAs and NF = −95 dB.

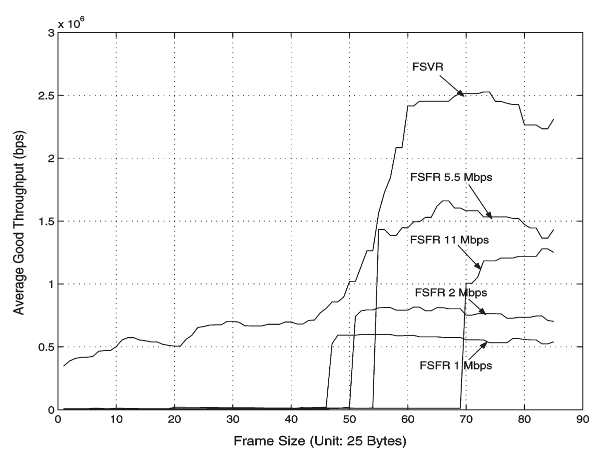

Fig. 14. Goodput performance of FSVR with five STAs and NF = −89 dB.

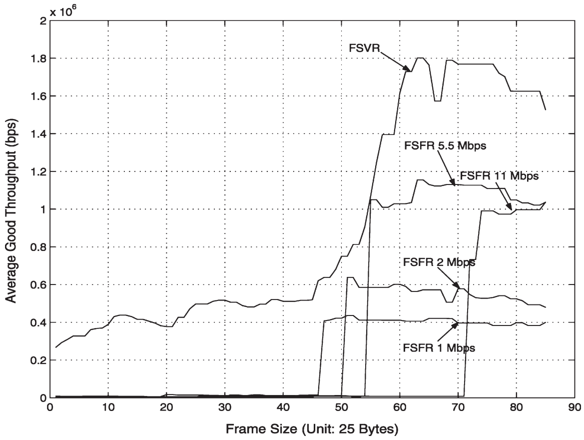

Fig. 15. Goodput performance of FSVR with 20 STAs and NF = −89 dB.

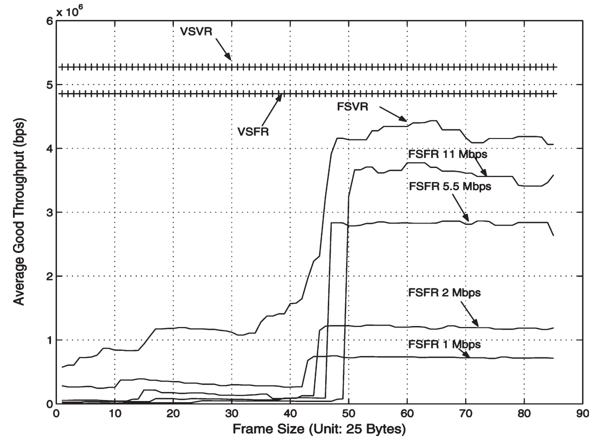

Fig. 16. Goodput performance of VSVR with five STAs and NF = −95 dB.

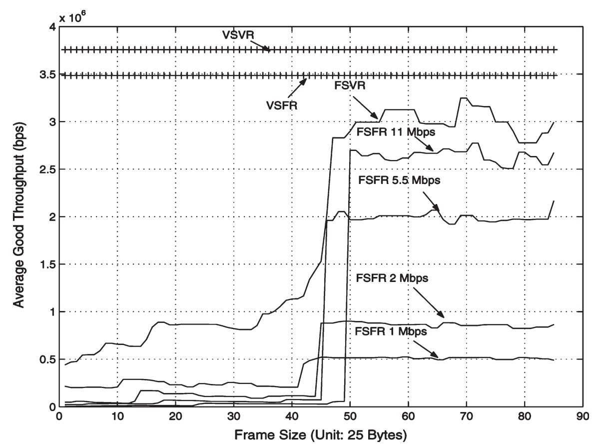

Fig. 17. Goodput performance of VSVR with 20 STAs and NF = −95 dB.

can achieve a better throughput than the FSFR scheme, as shown in Tables V and VI. However, the 'brick-wall' effect still exists; even though it is greatly mitigated.

The average goodput performance of the VSVR algorithm is shown in Figures 16, 17, 18 and 19. The VSVR performance is compared with the throughput performance achieved by the FSVR algorithm, the VSFR algorithm and the conventional FSFR scheme. We observe that using the VSVR algorithm achieves the best throughput than the proposed FSVR and VSFR algorithms. All proposed algorithms can achieve a better throughput performance than the conventional FSFR algorithm.

In this work, we also evaluated the performance of the frame size predictor based on the Kalman-filter

Table V. FSVR versus FSFR with L = 200 bytes, NF = −89 dB and M = 5.

Algorithm	Time(s)	Throughput (bps)	(95% confidence intervals)
FSFR (11 Mbps)	7.5986×10^3	1.0528×10^3	$(1.0395 \times 10^3, 1.0662 \times 10^3)$
FSFR (5.5 Mbps)	3.4858×10^3	2.2950×10^3	$(2.2303 \times 10^3, 2.3598 \times 10^3)$
FSFR (2 Mbps)	3.3613×10^3	2.3801×10^3	$(2.292 \times 10^3, 2.4681 \times 10^3)$
FSFR (1 Mbps)	1.0281×10^3	7.781×10^3	$(7.2413 \times 10^3, 8.3208 \times 10^3)$
FSVR	37.17	2.1211×10^5	$(2.0871 \times 10^5, 2.1551 \times 10^5)$

Table VI. FSVR versus FSFR with L = 200 bytes, NF = −95 dB and M = 20.

Algorithm	Time(s)	Throughput (bps)	(95% confidence intervals)
FSFR (11 Mbps)	2.0118×10^3	3.9765×10^3	$(3.4698 \times 10^3, 4.4833 \times 10^3)$
FSFR (5.5 Mbps)	4.5633×10^2	1.7531×10^4	$(1.6477 \times 10^4, 1.8585 \times 10^4)$
FSFR (2 Mbps)	1.9041×10^2	4.2015×10^4	$(3.435 \times 10^4, 4.9681 \times 10^4)$
FSFR (1 Mbps)	37.536	2.1313×10^5	$(2.088 \times 10^5, 2.1745 \times 10^5)$
FSVR	25.072	3.1908×10^5	$(3.1329 \times 10^5, 3.2486 \times 10^5)$

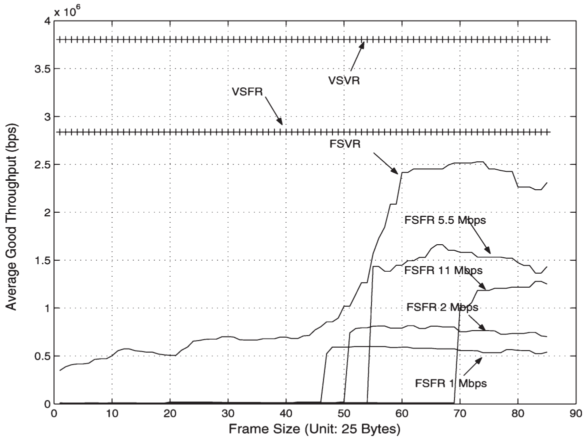

Fig. 18. Goodput performance of VSVR with five STAs and NF = −89 dB.

frame size predictor. The results are shown in Figures 20, 21, 22 and 23. We then integrated the optimal frame predictor with the proposed VSVR algorithm. Figures 24, 25, 26 and 27 show the results under different network scenarios.

Figure 24 shows that for five stations inside a service area, the goodput performance of the adaptive frame size scheme (VSFR), is compared with the fixed frame size scheme (FSFR) with the noise floor −89 dBm. Figure 25 is derived under the network scenario with five stations and a noise floor of −95 dBm. Figure 26 shows that the goodput performance of adaptive frame size scheme, is compared with the fixed frame size scheme with the noise floor −89 dBm and five stations inside a service area. Similarly, Figure 27 is the network scenario with five stations and a noise floor of −95 dBm.

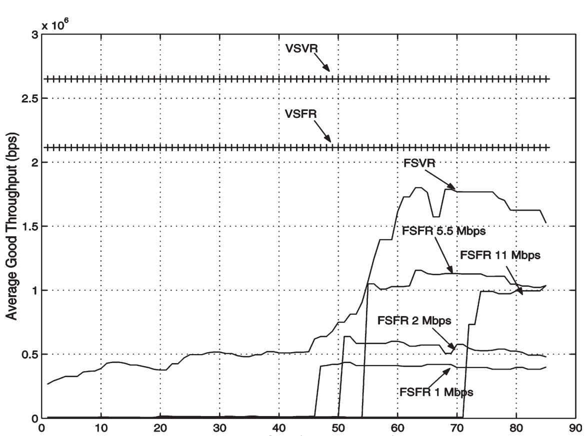

Fig. 19. Goodput performance of VSVR with 20 STAs and NF = −89 dB.

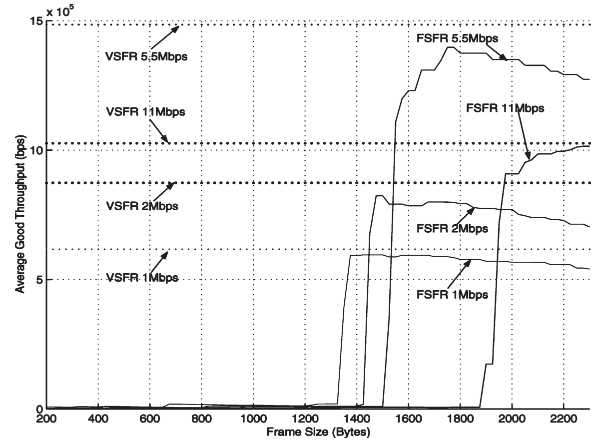

Fig. 20. Goodput performance of K-VSFR with five STAs and NF = −89 dBm.

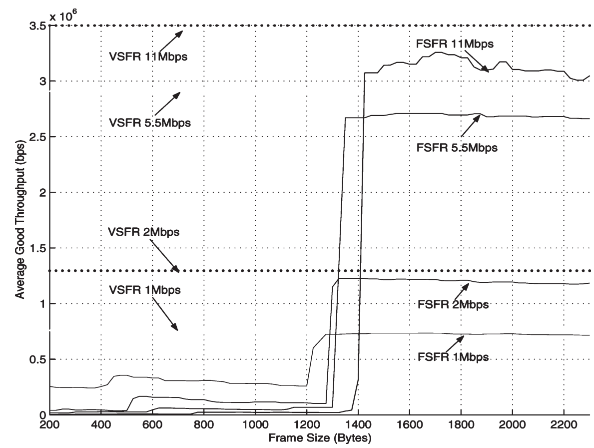

Fig. 21. Goodput performance of K-VSFR with five STAs and NF = −95 dBm.

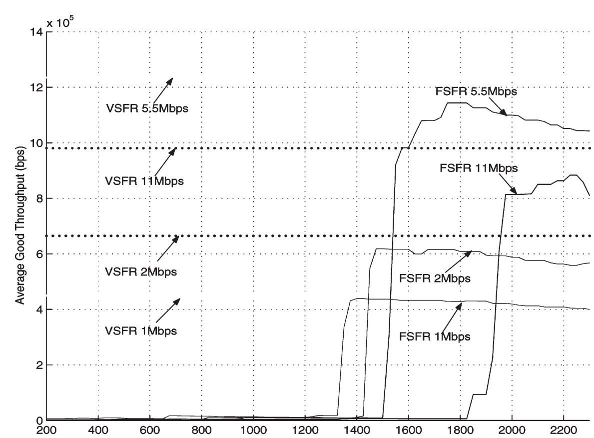

Fig. 22. Goodput performance of K-VSFR with 20 STAs and NF = −89 dBm.

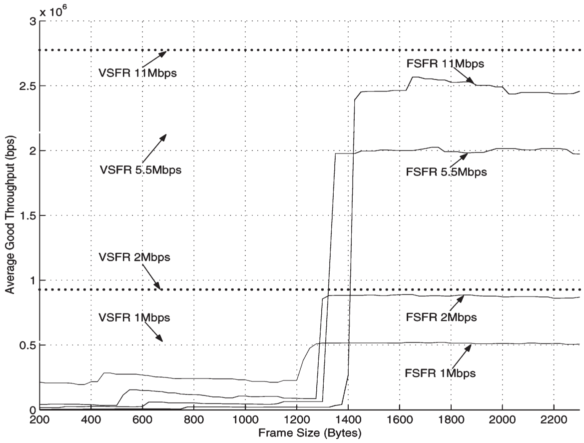

Fig. 23. Goodput performance of K-VSFR with 20 STAs and NF = −95 dBm.

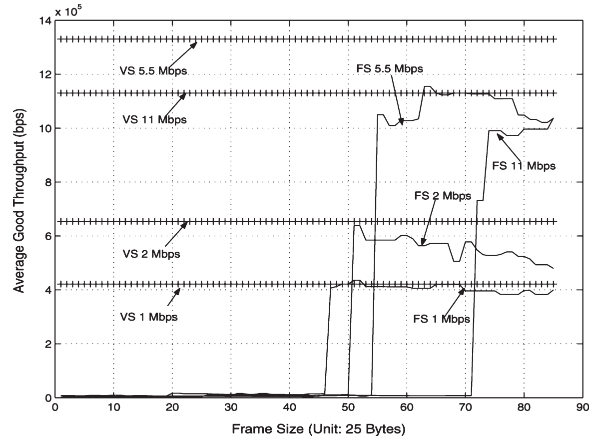

Fig. 26. Goodput performance of K-VSVR with 20 STAs and NF = −89 dBm.

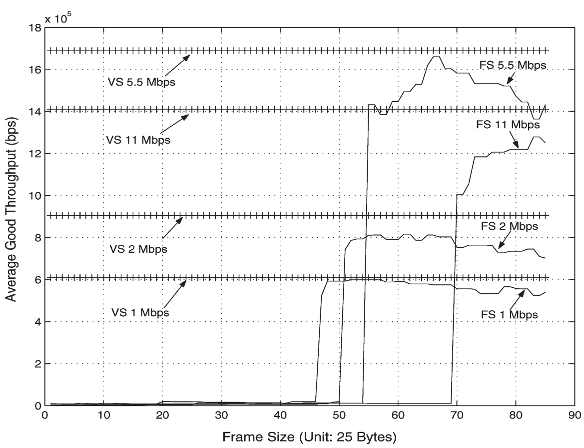

Fig. 24. Goodput performance of K-VSVR with five STAs and NF = −89 dBm.

From the above figures, we conclude that the proposed optimal frame size predictor based on the Kalman filter also achieves higher goodput than the traditional FSFR scheme, although the results are not as good as what is achieved by the maximum throughput predictor since the Kalman filter based frame size predictor is actually a data rate regulator. Moreover, when taking into account the effects of collisions and frame errors on the goodput performance, the system becomes non-linear, so the optimal frame size predictor based on the Kalman filter no longer provides accurate predictions. We should also notice the process model of the optimal frame size predictor based on the Kalman filter only takes frame errors into account without any consideration of the effects from protocol overhead and collisions.

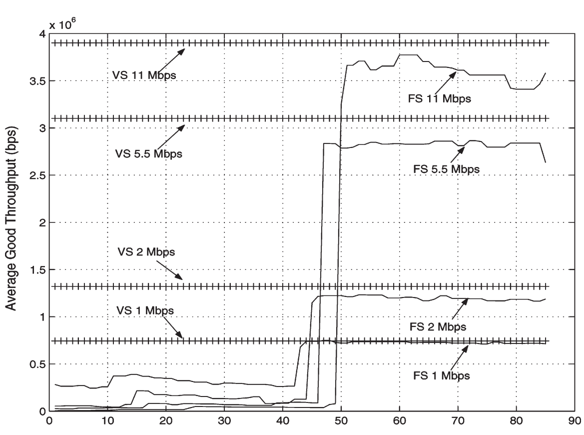

Fig. 25. Goodput performance of K-VSVR with five STAs and NF = −95 dBm.

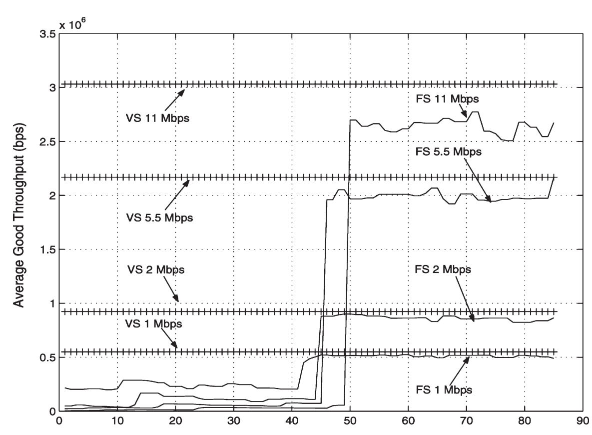

Fig. 27. Goodput performance of K-VSVR with 20 STAs and NF = −95 dBm.

7. Conclusions

Link adaptation is a very effective means to combat the time-varying wireless channel quality. In this paper, we have proposed a variable size and variable rate scheme for goodput improvement in the IEEE 802.11 wireless LAN. Since different applications may have various requirements on goodput performance, the goals for optimizing frame size may be different. Therefore, two optimal frame size predictors have been proposed in this work. An optimal frame size predictor based on the Kalman filter has been proposed to keep tracking variations in channel quality. This is to maintain a committed goodput. Another optimal frame size predictor has been studied by maximizing the goodput performance with considerations of collisions and frame errors for greedy applications. In addition, a data rate drafting scheme has been proposed for further goodput enhancement. We finally proposed a variable frame size and variable data rate (VSVR) algorithm. Simulation results show that the VSVR algorithm can double the channel throughput of current system implementations. Our study shows that the suggested solution at the MAC layer improves the QoS provisioning in wireless data networks and is an easy implementation for multimedia networks.

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