Performance of a Wireless Access Protocol on Correlated Rayleigh Fading Channels with Capture

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Abstract

The throughput performance of a wireless media access protocol taking into account the effect of correlated channel fading, capture, and propagation delay is analyzed. For efficient access on the uplink (mobile-to-base station link), the protocol makes use of the uplink channel status information which is conveyed to the mobiles through a busy/idle flag broadcast on the downlink (base station-to-mobile link). A first-order Markov model is used to describe the correlation in the packet success/failure process on a Rayleigh fading channel. The analytical results obtained through the first-order Markov approximation of the channel are compared to those obtained from an i.i.d. channel model. The Markovian fading channel model is shown to provide better performance results than the i.i.d. channel model. Simulations show that a first-order Markov approximation of the Rayleigh fading process is quite accurate. An enhanced version of the access protocol to take advantage of the memory in the fading channel behavior is proposed and analyzed. The effect of retransmission of erroneous data packets and propagation delay on the throughput is also analyzed. It is shown that the access protocol with an *error detect* feature is efficient in slow fading (e.g., pedestrian user speeds), whereas a *retransmission* protocol is more efficient in fast fading (e.g., vehicular user speeds).

Keywords: Wireless access, correlated fading, capture, Markov channel, retransmission.

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1 Introduction

Next generation wireless networks are envisaged to support high data rates, packet oriented transport, and multimedia traffic, and so the design and performance analysis of efficient media access protocols for integrated wireless multimedia applications are gaining importance [1], [2], [3]. One important issue that most analyses in the literature ignore is the effect of channel fading on the protocol performance. In fact, mobile radio channels are severely affected by time-varying losses due to distance, shadowing (blockage due to buildings, trees, etc.) and multipath fading. While the variation in the losses due to distance and shadowing is relatively slow, the variation due to multipath fading is quite rapid [4], [5]. The fading envelope due to multipath often follows a Rayleigh distribution, so that the envelope squared (i.e., the power) has an exponential distribution [5]. Most notably, the *correlation in the multipath fading* behavior and its effect on the performance of access protocols have not been adequately addressed in the literature so far. The primary focus of this paper is to address this void.

In the past, most models for data block transmission (e.g., in data link protocols) have assumed that the block transmissions were independent and identically distributed (i.i.d.). Also, many protocols were designed for an i.i.d. channel, and techniques were developed to eliminate channel memory (e.g., by interleaving). A newer approach is to take advantage of the channel correlation (e.g., exploiting some prediction techniques) to obtain better performance, rather than destroying the memory. A natural way to model a channel with memory is to approximate it by means of a Markov model. Markov models to approximate Rayleigh fading channels have been considered in [6], [7]. In [8], the binary process which describes packet successes and failures on a Rayleigh fading channel was investigated using mutual information, and it was shown that often a first-order Markov model is, in fact, an adequate approximation. In this paper, we adopt a first-order Markov approximation to the fading channel model in order to investigate the effect of Rayleigh fading on the performance of an access protocol. The protocol we consider in this paper makes use of the uplink (mobile-to-base station link) channel status information which is conveyed to the mobiles through the downlink (base station-to-mobile link) broadcast from the base station by periodically inserting a busy/idle flag [9]. This protocol can be viewed as a hybrid protocol employing the slotted ALOHA and the busy tone sensing concepts [10]. A header packet is sent on a contention basis first, following which data packets are sent on a reservation basis. By this approach, packet losses due to collision are restricted to occur only among header packet transmissions. Also, as we will see, capture conditions [11] are better handled in this protocol by allowing the base station to send a non-binary feedback to identify the successful mobile in the event of capture among simultaneous header packet transmissions from multiple mobiles.

The access protocol is described in Section 2. The fading channel model, packet success/failure process, and the parameters of the Markov approximation of the channel are presented in Section 3. The throughput performance of the protocol is analyzed in Section 4 assuming instantaneous and error-free feedback. We also propose and analyze an enhanced version of the protocol – called the protocol with an *error detect* (*ED*) feature – which attempts to improve the performance by exploiting the memory in the fading channel behavior. The strategy here is that a user, upon packet error events being detected in its ongoing message transmission, is advised by the base station to abort the transmission and attempt later. This scheme is expected to give good results in the presence of significant channel burstiness, as it avoids insisting on transmission in slots which are likely to be in error, and lets other users (whose channel conditions might be good) access the channel. However, in the presence of rapidly varying channels, which result in low correlation between errors in consecutive slots, the strategy of the *ED* protocol may be too wasteful, as it effectively reduces the message length and therefore decreases the overall efficiency. Another classic way of recovering errors in packet transmission is through *retransmission*. We analyze a parameterized retransmission strategy at the media access layer to recover erroneous data packets. In Section 5, we analyze the performance of these protocols when the feedback is not instantaneous, that is, when the delays due to propagation and processing are larger than the slot duration. Such scenarios are typical in high data rate wireless systems. Section 6 provides the conclusions.

2 Wireless Access Protocol

The operation of the wireless access protocol considered in this section (we will refer to it as the *basic* protocol) is described as follows. The uplink (mobile-to-base station link) channel is slotted to one packet duration. Transmission attempts are made by the mobiles only at the slot boundaries. Each message generated at the mobiles consists of two segments, namely, the *header segment* and the *data segment*. The header segment is one packet in length. It carries control information such as destination address, number of packets in the data segment, etc. The data segment represents the actual traffic. It consists of a random number of data packets. A *busy/idle flag* indicating the activity on the uplink is made available to the mobiles at the beginning of each slot. This flag is broadcast by the base station, once every slot, on the downlink (base station-to-mobile link).

According to the *basic* protocol, once a mobile receives a message to be delivered to the base station, it first checks the status of the received busy/idle flag. If the flag is set to busy, the mobile refrains from making a transmission attempt. If the flag is set to idle, then the mobile makes a transmission attempt by first sending the header packet on the uplink slot. If the header packet is received successfully (without packet loss due to collision or fading), the base station broadcasts the ID of the successful mobile (capturing mobile in the event of collision among header packets from different mobiles), and sets the flag to busy for the k subsequent slots, where k is the number of packets in the successful mobile's data segment. This allows only the successful mobile to send all its data packets continuously in those k slots. The base station resets the flag back to the idle status once the message transmission is completed. On the other hand, if the header packet is lost (due to collision or fading), the base station will not respond with a busy flag, but will continue sending the idle flag. This is an indication to the mobile that the header packet was lost, and so it has to reschedule the transmission attempt to a later time.

It can be seen that the packet transmissions, as per the above feedback mechanism (when error-free), can experience fading, interference, and noise during header transmission, whereas during data transmission only fading and noise (no interference) are experienced. Thus, in the case of error-free feedback to all mobiles, collisions and hence capture are possible only during the header packet transmission and not during the transmission of data packets. However, errors in the busy/idle flag reception would result in collisions, and hence packet losses, during the transmission of data packets as well.

3 Correlated Fading Channel Model

In order to analyze the performance of the access protocol in the presence of fading, we consider a frequency non-selective (flat) multipath fading channel, whose analytical model is described in the following. In the literature, the flat fading channel is modeled as a multiplicative complex function, $\alpha(t)$, which is adequately described as a random process. A popular model considers a complex Gaussian random process with a given mean and covariance function [5]. On the time scale of the fading variations, the process can be considered as stationary. Therefore, with no loss in generality, we will normalize its power to 1. The real and imaginary axes can be chosen so that the mean, $\mu = E[\alpha(t)]$, is real. Also, we consider the covariance function, defined as

$$K(\tau) = E[(\alpha(t+\tau) - \mu)^*(\alpha(t) - \mu)]. \tag{1}$$

Note that if $\mu = 0$, the envelope of $\alpha(t)$ is Rayleigh distributed for any t, and the envelope squared has an exponential distribution. When $\mu > 0$, the resulting fading envelope is Rician distributed and accounts for the presence of a line-of-sight (LOS) component. When the LOS component is absent, or has negligible power, the Rician model degenerates into the Rayleigh one.

In a widely accepted model, the Gaussian process is assumed to have a bandlimited non-rational spectrum given by [5]

$$S(f) = S(0)\left[1 - \left(\frac{f}{f_D}\right)^2\right]^{-1/2}, \quad \text{for } |f| < f_D, \tag{2}$$

and zero otherwise, where $f_D = V/\lambda$ is the Doppler bandwidth, V is the mobile speed, and λ is the carrier wavelength. This spectrum corresponds to the covariance function

$$K(\tau) = J_0(2\pi f_D|\tau|), \tag{3}$$

whose physical meaning has been investigated in [4], [5]. $J_0(\cdot)$ is the Bessel function of the first kind and of zeroth order. Note that the correlation properties of the fading process depend only on $f_D|\tau|$. When $f_D|\tau|$ is small (e.g., < 0.1), the process is very correlated ("slow" fading); on the other hand, for larger values of $f_D|\tau|$ (e.g., > 0.2), successive samples of the channel are almost independent ("fast" fading). For high data rates (i.e., small τ), the fading process can typically be considered as slowly varying, at least for the usual values of the carrier frequency (900-1800 MHz) and for typical mobile speeds, so that the dependence between transmissions of consecutive packets of data cannot be neglected. In particular, the assumption that the successes/failures of data packets constitute an i.i.d. process is far from reality, and may lead to incorrect results when used to evaluate the performance of wireless access protocols. Another approach, which accounts for dependence, is as follows.

The packet success/failure process is modeled as the outcome of a comparison of the instantaneous signal-to-noise ratio to a threshold value, SNR_t: if it is above the threshold, the packet is successfully decoded with probability 1; otherwise, the packet is lost with probability 1. If F is the value of the *fading margin*, the instantaneous signal-to-noise ratio (taking into account the effect of fading) is given by $SNR_t F|\alpha^2(t)|$. Hence, the binary process that describes packet successes/failures on the channel, β_j, can be obtained by quantization of the squared magnitude of the complex Gaussian description with the threshold $1/F$, i.e.,

$$\beta_j = \begin{cases} 0 & \text{if } v_j^2 > 1/F, \\ 1 & \text{if } v_j^2 \leq 1/F, \end{cases} \tag{4}$$

where $v_j = |\alpha(jT)|$ is the amplitude of the fading envelope at time jT, T is the packet duration, and "1" stands for a packet failure. We describe the above success/failure process on a mobile radio channel by a first-order Markov model. The parameters of the Markov model can be determined based on the fading model and the characteristics of the communications scheme. The transition probability matrix that describes the channel is given by

$$\boldsymbol{M}_c = \begin{pmatrix} p & 1-p \\ 1-q & q \end{pmatrix}, \tag{5}$$

where p and $1-q$ are the probabilities that the packet transmission in slot j is successful, given that the transmission in slot $j-1$ was successful or unsuccessful, respectively. Given the matrix $\boldsymbol{M}_c$, the channel model is completely characterized. In particular, the steady-state probability, P_E, that a packet error occurs due to fading and noise is

$$P_E = \frac{1-p}{2-p-q}. \tag{6}$$

Also, note that $(1-q)^{-1}$ represents the average length of a burst of errors, which is described by a geometric random variable. The parameters of the above Markov model can be found as [8]

$$P_E = 1 - e^{-1/F}, \tag{7}$$

and

$$q = 1 - \frac{Q(\theta, \rho\theta) - Q(\rho\theta, \theta)}{e^{1/F} - 1}, \tag{8}$$

where

$$\theta = \sqrt{\frac{2/F}{1-\rho^2}}, \tag{9}$$

$\rho = J_0(2\pi f_D T)$ is the correlation coefficient of two successive samples of the complex amplitude of a fading channel with Doppler frequency f_D, taken T seconds apart, $f_D T$ is the normalized Doppler bandwidth, and

$$Q(x,y) = \int_y^\infty e^{-\frac{(x^2+w^2)}{2}} I_0(xw) w\, dw \tag{10}$$

is the Marcum Q function. In (10), I_0 is the modified Bessel function of the first kind and of zeroth order. The Markov parameter p can be obtained using (8) and (7) in (6). By choosing different values of $f_D T$, we can establish fading channel models with different degrees of correlation in the fading process.

3.1 Header Packet Capture Probability

Header packets from different mobiles can collide in a slot and the probability of capture under such conditions needs to be computed to carry out the throughput analysis. Consider simultaneous header packet transmissions from n different mobiles in a slot, where the signal from user j is received at the base station with power α_j^2, $j = 1, 2, .., n$. As stated earlier, in a multipath fading environment and in the absence of a LOS component, the α_j's are Rayleigh distributed and the α_j^2's are exponentially distributed. Capture is said to occur in favor of user i if

$$\alpha_i^2 > B\left[\sum_{j=1, j\neq i}^{n} \alpha_j^2 + N_0\right] = B\left[\sum_{j=1, j\neq i}^{n} \alpha_j^2\right] + \frac{1}{F}, \tag{11}$$

where B is defined as the capture threshold and N_0 is the noise power[1]. From [12], the probability that there is a header packet success when n simultaneously colliding header packet transmissions are present in a slot, $p_s^{(n)}$, can be found as

$$p_s^{(n)} = ne^{-1/F}\left(\frac{1}{1+B}\right)^{n-1}. \tag{12}$$

From (12), there is no capture when $B \to \infty$ (i.e., $p_s^{(n)} = 0$ for $n > 1$). Also, $B = 1$ corresponds to an idealized perfect capture condition [11].

4 Throughput Analysis

To analyze the system throughput, we initially assume that the feedback from the base station (busy/idle flag and the successful mobile ID) on the downlink is received instantaneously and error-free by all mobiles. In the subsequent analysis in Section 5, we will relax the instantaneous feedback assumption. The error-free feedback assumption is reasonable because the feedback consists of only a few bits, which can be provided with adequate error protection. We also make the following assumptions on the message arrival process and the message length distribution: $1)$ the message arrival process at each mobile is Bernoulli with rate λ per slot (i.e., there are no arrivals or one arrival with probabilities $1-\lambda$ and λ, respectively), and $2)$ the length of the data segment of the message (not including the header packet), k, measured in integer number of packets, follows a geometric distribution with parameter g_d ($0 < g_d < 1$), and probability mass function

$$P[k = y] = \begin{cases} g_d(1-g_d)^{y-1} & y = 1, 2, 3, ... \\ 0 & \text{otherwise.} \end{cases} \tag{13}$$

For the Bernoulli arrival process, the probability that n out of N mobile users ($0 \leq n \leq N$) make a transmission attempt in a slot, p_n, is given by

$$p_n = \binom{N}{n}\lambda^n(1-\lambda)^{N-n}. \tag{14}$$

[1]Note that, since in our model all powers are normalized so that $E[\alpha_i^2] = 1$, $1/N_0$ is the average signal-to-noise ratio and $1/(BN_0)$ is the uplink fading margin.

We first analyze the performance of the *basic* protocol described in Section 2. Later, we extend the analysis to both an enhanced version of the protocol with an *error detect* feature that attempts to exploit the channel memory for better performance, as well as a *retransmission* protocol to recover erroneous data packets.

4.1 Basic Protocol Performance

Based on the assumptions described above, the evolution of the *basic* protocol on a Markovian channel can be tracked by means of a Markov chain with a finite number of states. An adequate state space consists of just five states describing the system status in a slot, namely: $1)$ Idle (I), $2)$ Header packet success (H_s), $3)$ Header packet failure (H_f), $4)$ Data packet success (D_s), and $5)$ Data packet failure (D_f). Once the state transition probabilities of the chain, $P_{ij},\ i, j \in \Omega$, where Ω is the state space, are determined, the steady-state probability vector, π, is given by the solution of the equations

$$\pi = \pi \boldsymbol{P}, \qquad \sum_{j \in \Omega} \pi_j = 1. \tag{15}$$

The state transition probability matrix for the *basic* protocol, $\boldsymbol{P}_{basic}$, can be written as

$$\boldsymbol{P}_{basic} = \begin{bmatrix} X_0 & X_1 & X_2 & 0 & 0 \\ 0 & 0 & 0 & p' & 1-p' \\ X_0 & X_1 & X_2 & 0 & 0 \\ g_d X_0 & g_d X_1 & g_d X_2 & X_3 & X_4 \\ g_d X_0 & g_d X_1 & g_d X_2 & X_5 & X_6 \end{bmatrix}, \tag{16}$$

where $X_0 = p_0 = (1-\lambda)^N$, $X_1 = \sum_{i=1}^{N} p_i p_s^{(i)} = (1-P_E)p_1(1+x)^{N-1}$, $x = \frac{\lambda}{(1+B)(1-\lambda)}$, $X_2 = 1-X_0-X_1$, $X_3 = (1-g_d)p$, $X_4 = (1-g_d)(1-p)$, $X_5 = (1-g_d)(1-q)$, and $X_6 = (1-g_d)q$. The arrival probabilities p_i, and the capture probabilities $p_s^{(i)}$, are given in (14) and (12), respectively. Note that in order to precisely track the transitions from Header success state (H_s) to Data success state (D_s) one would need to increase the number of states to take into account all the possible numbers of simultaneously transmitting users in the Header success state, on which the probability of having a data packet success in the next slot depends. Here, in order to have a smaller number of states, we assume that a transition from Header success state to Data success state occurs with probability p'. In particular, we will consider $p' = 1$ as an optimistic approximation, and $p' = p$ as a pessimistic approximation. We show that the throughput estimates obtained from these two approximations are tight, and they closely agree with the actual throughput values obtained by simulation. In all numerical results, we assume $p' = p$ so that the results presented are to be considered as conservative estimates.

Working through the steady state analysis of the system, we obtain the expression for the average throughput of the *basic* protocol, in closed-form, as

$$\eta_{basic} = \frac{X_1}{g_d + X_1} \left[\frac{g_d p' + (1-g_d)(1-q)}{1 + (1-g_d)(1-p-q)} \right]. \tag{17}$$

Note that, since $X_1/(g_d + X_1)$ is a monotonically increasing function of X_1, and since X_1 is the only quantity in (17) which depends on λ, the value of λ which maximizes X_1 also maximizes the throughput. In fact, the maximum throughput, η_{max}, occurs at the arrival rate, λ_{max}, given by

$$\lambda_{max} = \frac{1+B}{BN}, \tag{18}$$

which is independent of p, q and g_d. Note that when there is no capture (i.e., $B \to \infty$), λ_{max} becomes equal to $1/N$. When there is perfect capture (i.e., $B = 1$), λ_{max} becomes equal to $2/N$. In (17), successful header packet transmissions are not considered to be useful in the throughput computation, so that $\eta_{basic} = \pi_4$, where π_4 is the steady-state probability of state D_s. The protocol throughput for an i.i.d. channel model can be derived by setting $p = p' = 1 - q = 1 - P_E$ in (17), to obtain

$$\eta_{iid} = \frac{X_1}{g_d + X_1}(1 - P_E). \tag{19}$$

4.2 Enhanced Protocol with Error-Detect Feature

In the *basic* protocol described and analyzed above, we allowed the mobile to continuously transmit all the packets in the data segment of the message even when one or more of those packets were lost due to channel fading. However, the memory in the fading process can be exploited to modify the data transmission strategy by using the knowledge about the channel status information. As an example, consider the following. Under slow fading conditions (where events in successive slots are expected to be highly correlated), the fact that the data packet in the current slot is received in error implies that the packet in the subsequent slot will also be received in error with high probability. Therefore, the protocol rules can be modified so that, when the base station detects such a "bad" channel condition during transmission of a data segment from a mobile, it could ask that mobile to abort transmission and release the channel. This avoids the likely occurrence of subsequent errors, and allows other mobiles (which may, on the other hand, experience "good" channel conditions) to transmit.

To investigate how this idea can be used to enhance the protocol performance, we analyze a simple, modified version of the *basic* protocol. We refer to the modified scheme as the protocol with an *error detect* (*ED*) feature. According to the *ED* protocol, if a packet in the data segment of the message is received in error, the base station sends out an idle flag in the next slot (instead of sending a busy flag in all k data slots, as the basic protocol would do) to prompt the mobile to terminate the ongoing data transmission. Such a strategy enables other mobiles to access the channel during those slots which otherwise could have witnessed, with high probability, loss of packets due to fading.

Note that the transition probability matrix for the *ED* protocol, $\boldsymbol{P}_{ED}$, will be the same as that of the *basic* protocol (matrix $\boldsymbol{P}_{basic}$, given by (16)), except for the transition probabilities from state D_f. In fact, for the *ED* protocol, the transition probabilities from state D_f will be the same as those from the idle state.

Accordingly, the transition probability matrix for the *ED* protocol is given by

$$\boldsymbol{P}_{ED} = \begin{bmatrix} X_0 & X_1 & X_2 & 0 & 0 \\ 0 & 0 & 0 & p' & 1-p' \\ X_0 & X_1 & X_2 & 0 & 0 \\ g_d X_0 & g_d X_1 & g_d X_2 & X_3 & X_4 \\ X_0 & X_1 & X_2 & 0 & 0 \end{bmatrix}, \tag{20}$$

where X_0, X_1, X_2, X_3, and X_4 are as defined in Section 4.1. From (20), the throughput expression for the *ED* protocol can be derived as

$$\eta_{ED} = \frac{p' X_1}{1-(1-g_d)p + X_1[1-(1-g_d)(p-p')]}, \tag{21}$$

which, again, is maximum for $\lambda = \frac{1+B}{BN}$.

4.3 Protocol with Retransmission of Erroneous Data Packets

The *basic* protocol does not take any action in the event of data packet errors, i.e., data packets which are corrupted during transmission are just lost and the recovery of such errors are left to higher layer protocols. The *ED* protocol described in the previous subsection, on the other hand, reacts to packet errors by aborting the ongoing message transmission. However, in the presence of rapidly varying channels, which result in low correlation between errors in consecutive slots, the strategy of the *ED* protocol may be too wasteful, as it effectively reduces the message length and therefore decreases the overall efficiency. Another classic way of recovering errors in packet transmission is through *retransmission*. Instead of ignoring packet errors (as in the *basic* protocol) or aborting the message transmission altogether (as in the *ED* variation), a packet is retransmitted if it is received in error. In the local wireless environment under consideration, where the feedback is assumed to be instantaneous, a packet in error can be retransmitted in the immediately following slot. In this case, the base station would need to send a non-binary feedback (busy/idle/retransmit) in order to avoid a collision among a retransmission packet and header packets from other users. Thus, with the retransmission strategy, a geometric length message of L packets (with $E[L] = 1/g_d$) will take L' slots to finally get through, due to possible retransmissions. Therefore, we will have

$$L' = \sum_{j=1}^{L} Y_j, \tag{22}$$

where Y_j is an integer random variable equal to the number of transmissions it takes packet j to be successfully received. Note that the first transmission of data packet j has a probability of success p for $j > 1$. This is because the first transmission of packet j immediately follows the successful transmission of packet $j-1$. On the other hand, if the transmission is unsuccessful, then the probability that the immediately following retransmission succeeds is $1-q$. Thus, for $j > 1$, the random variable Y_j has the following probability mass function:

$$P[Y_j = y] = \begin{cases} p & y = 1 \\ (1-p)q^{y-2}(1-q) & y > 1. \end{cases} \tag{23}$$

The same argument applies for $j = 1$ as well, but the probability p' is to be used instead of p in this case.

Since the use of retransmissions occurs only when a data packet is in error, the transition matrix for the protocol is the same as for the *basic* protocol, except for the last row (transitions from state D_f, which corresponds to a data packet in error). In fact, after an erroneous data packet, a retransmission attempt is always performed and therefore only transitions to D_s (with probability $1 - q$) or D_f (with probability q) are allowed (in other words, a message cannot end with a failed transmission). Therefore, the transition probability matrix for the *retransmission* protocol can be written as

$$\boldsymbol{P}_{retx} = \begin{bmatrix} X_0 & X_1 & X_2 & 0 & 0 \\ 0 & 0 & 0 & p' & 1-p' \\ X_0 & X_1 & X_2 & 0 & 0 \\ g_d X_0 & g_d X_1 & g_d X_2 & X_3 & X_4 \\ 0 & 0 & 0 & 1-q & q \end{bmatrix}. \tag{24}$$

From the above, the throughput expression for the *retransmission* protocol can be derived as

$$\eta_{retx} = \frac{(1-q)X_1}{g_d(1-q) + X_1[2 - p - q + g_d(p - p')]}, \tag{25}$$

which also results in maximum throughput when $\lambda = \frac{1+B}{BN}$. In the case of $p' = p$, (25) can be further simplified as

$$\eta_{retx} = \frac{p_1(1 - P_E)(1+x)^{N-1}}{g_d + p_1(1+x)^{N-1}}, \tag{26}$$

where $x = \frac{\lambda}{(1+B)(1-\lambda)}$, and p_1 is obtained from (14) for $n = 1$. From (26), it can be seen that the *retransmission* protocol throughput remains independent of the Markov parameters of the channel, p and q.

Further, instead of terminating the data transmission at the first instance of a data packet failure (as in the *ED* protocol), or repeatedly sending a data packet until success (as in the *retransmission* protocol), the base station could allow the mobile to resend a lost data packet only a limited number of times (defined as a parameter, n_r), after which the mobile is asked to abort the data transmission. As can be seen, this is a generalized form of the protocol, and both *ED* and *retransmission* protocols can be thought of as special cases of this generalized form for $n_r = 0$ and $n_r \to \infty$, respectively. The transition probabilities for the generalized protocol can be written exactly as for the *ED* protocol, except that the single D_f state in the *ED* case is expanded into $(n_r + 1)$ different states, that is, $D_f(0), D_f(1), ..., D_f(n_r)$, where $D_f(j)$ corresponds to failure of the $(j+1)^{th}$ transmission of a data packet. Accordingly, the state transition probability matrix for the parametrized retransmission strategy can be written as

$$\boldsymbol{P}'_{retx} = \begin{bmatrix} X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & . & . & 0 \\ 0 & 0 & 0 & p' & 1-p' & 0 & 0 & . & . & 0 \\ X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & . & . & 0 \\ g_d X_0 & g_d X_1 & g_d X_2 & X_3 & X_4 & 0 & 0 & . & . & 0 \\ 0 & 0 & 0 & 1-q & 0 & q & 0 & . & . & 0 \\ 0 & 0 & 0 & 1-q & 0 & 0 & q & . & . & 0 \\ . & . & . & . & . & . & . & . & . & . \\ . & . & . & . & . & . & . & . & . & . \\ 0 & 0 & 0 & 1-q & 0 & 0 & 0 & . & . & q \\ X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & . & . & 0 \end{bmatrix}. \tag{27}$$

4.4 Results and Discussion

In Figure 1, the throughput performance of the different versions of the access protocol obtained from (17), (19), (21), and (25) is plotted versus the message arrival rate, λ, under no capture condition (i.e., $B \to \infty$) and for $N = 10$ users. A g_d value of 0.1, corresponding to an average message length of 10 packets per message (not including the header) is used. Plots are shown for uplink fading margins, F, of 5 dB and 10 dB. For the *basic*, *ED*, and *retransmission* protocols, a normalized Doppler bandwidth, $f_D T$, of 0.02 (representing slow fading) is chosen. The extreme case of i.i.d. packet errors with the same marginal error rate is also plotted for comparison. The effect of varying the normalized values of the Doppler bandwidth, $f_D T$, the uplink fading margin, F, and the average message length, $1/g_d$, is illustrated in the subsequent graphs (Figures 2, 3, and 4) for the case of no capture. In Figures 1 through 4, the solid lines represent the analytical results obtained from the throughput expressions, whereas the markers represent simulation points. As mentioned earlier, in computing the analytical results, we computed the throughput by choosing $p' = p$. This is a good approximation for all the protocols considered because, over a range of values of different parameters, the estimated throughputs were found to be at most 2 to 2.5% worse than optimistic throughputs obtained by using $p' = 1$. Further, the results obtained by explicit simulation of the protocols on a correlated fading channel, are seen to be in close agreement with the analytical results. The correlated Rayleigh fading channel was simulated using the method proposed by Jakes [5].

From Figure 1, we observe the following. As was analytically computed, for all three protocol schemes (*basic/ED/retransmission*) and both channel models (slow fading/i.i.d.), the maximum achievable throughput occurs when $\lambda = 1/N$, and this gives good robustness against either possible variations or wrong estimates of the channel parameters. Even with a small uplink fading margin of 5 dB, the *basic* protocol is found to offer a maximum throughput of about 0.631 successful data packets per slot. The *ED* protocol is found to perform better than both the *basic* and the *retransmission* protocols (e.g., maximum throughput of 0.673 for the *ED* protocol against 0.631 for the *basic* protocol, and 0.579 for the *retransmission* protocol). This was to be expected, because the fading rate considered is small (i.e., $f_D T$ = 0.02), and the *basic* protocol allows the mobiles to transmit all the data packets in a message without any break, even under deep fade situations. On the other hand, if a mobile encounters a deep fade during its data segment transmission, the *ED* protocol releases the channel from that mobile and allows other mobiles to access the channel, thereby increasing the channel utilization efficiency. Further, for the considered $f_D T$ value of 0.02, it is also reasonable for the *retransmission* protocol to perform poorer than both the *basic* and the *ED* protocols, because the high burstiness of the packet errors at $f_D T = 0.02$ would cause the *retransmission* protocol to use more slots for resending the erroneous packets. However, at high values of $f_D T$, as seen from Figure 2, the *retransmission* protocol performs better than *basic* and *ED* protocols. When the uplink fading margin is 10 dB, the relative performance of all the three protocols remains the same, but the actual differences in performance become smaller. For example, the maximum throughput for the *basic*, *ED*, and *retransmission* protocols are 0.728, 0.742, and 0.704, respectively.

A comparison between the performance predictions of an i.i.d. channel model and the Markov fading model is also made in Figure 1. It is seen that the i.i.d. model provides a pessimistic estimate of the throughput performance compared to the Markov model under slow fading conditions (i.e., $f_D T$ = 0.02). However, it has been found that for fast fading conditions (e.g., $f_D T$ = 1), both the i.i.d. and Markov models tend to produce close performance results. It is further noted that the first-order Markov approximation of the fading process is quite accurate, since the results computed through Markov analysis closely agree with those obtained by direct simulation of the protocol and the fading process.

Figure 2 illustrates the effect of varying $f_D T$ on the maximum achievable throughput for the *basic*, *ED*, and *retransmission* protocols at $g_d = 0.1$, $N = 10$, $B \to \infty$, and $\lambda = 1/N$. As seen earlier, the *ED* protocol performs better than both the *basic* and the *retransmission* protocols for small values of $f_D T$ (e.g., < 0.08 for $F = 5$ dB), whereas at high values of $f_D T$ (i.e., for weaker correlation between packet errors), the *retransmission* protocol performs better than the *basic* and *ED* protocols. This performance crossover is due to the fact that at high values of $f_D T$, each packet during the data segment transmission would experience a nearly i.i.d. success/failure event, and terminating the data transmission in the event of a single packet failure (as done in the *ED* protocol) reduces the average number of packet success events during the data segment transmission. This observation suggests that the *ED* protocol is suited for networks supporting many slowly moving users (e.g., local area environment), and the *retransmission* protocol is more suited for networks supporting many fast moving users (e.g., cellular environment).

Figure 3 shows the effect of varying the uplink fading margin, F, on the throughput performance of the protocols when $N = 10$, $g_d = 0.1$, $B \to \infty$, $\lambda = 1/N$, and $f_D T = 0.02$. At low fading margins, the *ED* protocol performs markedly better than the other protocols. At high fading margins (e.g., $>$ 15 dB), all the protocols tend to perform almost the same. However, over a typical range of uplink fading margins (5 dB - 10 dB), the *ED* protocol shows noticeable improvements over the *basic* and *retransmission* protocols under slow fading conditions. Figure 4 shows the effect of increasing the average message length, $1/g_d$, on the throughput. It can be seen that the throughput improves for smaller values of g_d, i.e., for longer message lengths. This suggests that the protocol is suitable for messaging applications like file transfers, etc. However, the increased throughput for large message sizes will come at the expense of increased delay performance of the protocols [13].

The effect of header packet capture on the maximum throughput performance of all the protocols is illustrated in Figure 5 for fading margins of 5 and 10 dB, at an $f_D T$ value of 0.02. In the case of the *basic* protocol with 5 dB fading margin, the maximum achievable throughput increases from 0.631 under no capture (e.g., $B = 30$ dB) to 0.726 under perfect capture (i.e., $B = 0$ dB). This is about 15% increase in the maximum achievable throughput due to the header packet capture phenomenon. For the same set of conditions, the *ED* protocol resulted in an 18% increase (0.673 to 0.795) in maximum throughput, and the *retransmission* protocol resulted in an increase of about 11% (0.579 to 0.645).

In Figure 6, maximum throughput performance curves are plotted for the *parameterized retransmission* strategy as a function of $f_D T$ for no capture. These curves are obtained by solving (27) for different values of the parameter n_r (i.e., 1, 2, 5, 10). The performance of *ED* and *retransmission* protocols are also plotted as limiting cases when $n_r = 0$, and $n_r \to \infty$, respectively. As observed earlier, the *ED* protocol performs best when the fading is slow, and worst when fading is fast. Under fast fading conditions (e.g., $f_D T > 0.1$), it is interesting to see that the performance improves significantly compared to the *ED* protocol even if only one retransmission ($n_r = 1$) is allowed for the lost data packet recovery. In fact, just 2 or 3 retransmissions are adequate to establish almost the same performance as the *retransmission* protocol under fast fading conditions. Even in slow fading ($f_D T < 0.1$), the *parameterized retransmission* strategy performs well, close to the *ED* protocol's performance which is best in slow fading. In summary, a *parameterized retransmission* strategy which allows a maximum of 2 or 3 retransmissions is found to result in good performance over the range of mobile speeds of interest.

We further note here that in all the analyses presented above, it has been assumed that when a header packet error occurs, the header is simply dropped and the message is regenerated as a new arrival according to a Bernoulli process. A more comprehensive analytical model which considers retransmission of lost header packets by allowing the mobile users to be either in a *backlogged* state or *non-backlogged* state is studied in [13]. The results in [13] are qualitatively similar to those presented in this paper.

5 Effect of Propagation and Processing Delays

The instantaneous feedback (i.e., zero propagation and processing delay) assumed in the previous section can be valid in situations where the delays due to propagation and processing are very small compared to the slot duration. However, this assumption may not be always valid, particularly when high data rate transmissions are considered. In this section, we analyze the performance of the *basic*, *ED*, and *retransmission* protocols when the feedback is not instantaneous. Analyzing the protocol performance for any general value of delay, when it spans multiple slot intervals, appears difficult (the state space quickly becomes too large). However, in local wireless environments propagation delays are small so that consideration of propagation and processing delays spanning at most one slot interval seems adequate. Here, we derive approximate expressions for the throughput performance when the propagation and processing delay span *one slot interval*. Simulations show that the approximate analytical predictions are fairly accurate.

In the case of non-zero propagation and processing delay, following a header packet transmission, the mobile waits for the busy/idle feedback from the base station. It does not send any data packet in the slots that elapse between the header transmission and the receipt of the feedback. These *vulnerable slots* are sensed as idle by other mobiles which may send their header packets in those slots. However, if the feedback received immediately following the vulnerable slot(s) is a busy flag along with the successful mobile ID, the mobiles which sent header packets in the vulnerable slot(s) will stop and attempt later. The successful mobile will then send the data packets following the rules of the particular protocol. In the case of the *basic* protocol, the base station would set the k slots following the vulnerable slot(s) as busy, over which the mobile sends all its k data packets. In the case of the *ED* protocol, if the base station encounters a packet error in the ongoing data transmission, it sends out an idle flag, thus requiring the mobile to terminate the data transmission. However, because of the delay, there will be a time lag (equal to the number of vulnerable slots) in the actual termination of data transmission at the mobile. In the case of the *retransmission* protocol, for each data packet that was received in error, the base station sends out a retransmit flag to enable recovery of those lost data packets. In addition, the base station has to *force* the slot next to the last data packet (i.e., the k^{th} data packet, where k is the number of packets in the data segment) transmission slot to go unused (i.e., nobody is allowed to send header packets in that slot). This has to be done because of the time lag experienced by the mobile in knowing whether it has to retransmit the last data packet or not. Thus, in the *retransmission* protocol, the slot following the successful last data packet slot is always wasted.

In the following, we provide the throughput analysis for the case where the number of vulnerable slots is one. To make the analysis tractable, we make the following simplifying assumption. Even if there is no header success in the slot prior to a vulnerable slot, the other mobiles which sent header packets in the vulnerable slot are instructed by the base station to stop immediately and postpone their attempts to a later time, thus ignoring a possible header success in the vulnerable slot. By forcing this constraint, the slot following the vulnerable slot is either used for data packet transmission by a mobile whose header succeeded in the slot prior to the vulnerable slot, or available as an idle slot for all the mobiles to contend.

With the above assumption, the following seven different states are adequate to represent the state space for the *basic* Protocol: 1) Idle, 2) Header packet success, 3) Header packet failure, 4) Vulnerable slot following a header success, 5) Vulnerable slot following a header failure, 6) Data packet success, and 7) Data packet failure. From the Header packet success state (State 2), the system moves to State 4 with probability 1. Likewise, the system moves from State 3 to State 5 with probability 1. The probability of a data packet success/failure in the slot next to the vulnerable slot is obtained as the two-step transition probability from the the slot in which the header was sent, given by the appropriate entry of $\boldsymbol{M}_c^2$. In other words, from State 4, the system moves to States 6 and 7 with probabilities $X_7 = p^2 + (1-p)(1-q)$, and $X_8 = p(1-p) + (1-p)q$, respectively. Note that state 4 does not specify the channel status in the vulnerable slot, so that the most recent information available about the channel relates to the previous slot. Thus, the state transition probability matrix for the *basic* protocol with one slot propagation and processing delay, $P_{basic}^{(d)}$, can be written as

$$\boldsymbol{P}_{basic}^{(d)} = \begin{bmatrix} X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 1 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & X_7 & X_8 \\ X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 \\ g_d X_0 & g_d X_1 & g_d X_2 & 0 & 0 & X_3 & X_4 \\ g_d X_0 & g_d X_1 & g_d X_2 & 0 & 0 & X_5 & X_6 \end{bmatrix}, \tag{28}$$

and the throughput is obtained as

$$\eta_{basic}^{(d)} = \pi_6. \tag{29}$$

In the case of the *ED* protocol, States 1 to 5 remain the same as States 1 to 5 in the *basic* protocol state space. However, because of the termination of data transmission following a data packet failure, the Data packet success state and the Data packet failure state in the *basic* protocol are to be expanded into four different states, namely, Data packet success state following a data packet failure (State 6), Data packet success state not following a data packet failure (State 7), Data packet failure state following a data packet failure (State 8), and Data packet failure state not following a data packet failure (State 9). Thus, a total of nine different states are needed to adequately describe the *ED* protocol state space. The state transition probability matrix for the *ED* protocol with one slot propagation and processing delay, $P_{ED}^{(d)}$, can be written as

$$\boldsymbol{P}_{ED}^{(d)} = \begin{bmatrix} X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & X_7 & 0 & X_8 \\ X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & 0 & 0 \\ X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & 0 & 0 \\ g_d X_0 & g_d X_1 & g_d X_2 & 0 & 0 & 0 & X_3 & 0 & X_4 \\ X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & 0 & 0 \\ g_d X_0 & g_d X_1 & g_d X_2 & 0 & 0 & X_5 & 0 & X_6 & 0 \end{bmatrix}, \tag{30}$$

and the throughput of the *ED* protocol is obtained as

$$\eta_{ED}^{(d)} = \pi_6 + \pi_7. \tag{31}$$

Finally, twelve different states are needed to describe the *retransmission* protocol state space. Because the slot next to the successful last data packet slot is forcefully wasted in the *retransmission* protocol, the idle state is expanded into two separate states, namely, Idle state following a data packet failure (State 1), and Idle state not following a data packet failure (State 2). States 3 to 6 in the *retransmission* protocol state space are the same as States 2 to 5 of the *basic* and *ED* protocols state space. Also, the Data success and Data failure states in the *basic* protocol need to be expanded into six different states, namely, Data packet success state a) following a data packet failure (State 7), b) following an idle (State 8) and c) following neither data packet failure nor idle (State 9), and Data packet failure state a) following a data packet failure (State 10), b) following an idle (State 11) and c) following neither data packet failure nor idle (State 12). With these twelve different states, the state transition probability matrix for the *retransmission* protocol with one slot propagation and processing delay, $P_{retx}^{(d)}$, can be written as

$$\boldsymbol{P}_{retx}^{(d)} = \begin{bmatrix} 0 & 0 & 0 & 0 & 0 & 0 & 0 & X_9 & 0 & 0 & X_{10} & 0 \\ 0 & X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & X_7 & 0 & 0 & X_8 \\ 0 & X_0 & X_1 & X_2 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & p & 0 & 0 & 1-p \\ 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & g_d & 0 & 0 & 0 & 0 & 0 & 0 & X_3 & 0 & 0 & X_4 \\ 0 & 0 & 0 & 0 & 0 & 0 & 1-q & 0 & 0 & q & 0 & 0 \\ 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ g_d & 0 & 0 & 0 & 0 & 0 & X_5 & 0 & 0 & X_6 & 0 & 0 \end{bmatrix}, \tag{32}$$

where $X_9 = q(1-q) + (1-q)p$, and $X_{10} = q^2 + (1-q)(1-p)$. The throughput of the *retransmission* protocol is obtained as

$$\eta_{retx}^{(d)} = \pi_7 + \pi_8 + \pi_9. \tag{33}$$

The throughput performance of the *basic*, *ED*, and the *retransmission* protocols with one slot propagation and processing delay as computed by solving (28), (30), and (32) are plotted in Figure 7. The parameters used for these plots are $N = 10$, $g_d = 0.1$, $F = 5$ dB, $B \to \infty$, and $f_D T = 0.02$. The results generated by explicit simulations without any simplifying assumption are also plotted (in markers). In the analysis, header success events in vulnerable slots are ignored, whereas in the simulations the event of a header success in the vulnerable slot and subsequent data packet transmission is allowed. It is seen from Figure 7 that both analysis and simulation results agree very well for all three protocols. A comparison of the throughput performance of the three protocols for zero and one slot propagation and processing delay is provided in Figures 8 (for slow fading, $f_D T = 0.02$) and Figure 9 (for fast fading, $f_D T = 0.64$). One slot propagation and processing delay case is typically found to result in about 13% to 20% degradation in the maximum throughput compared to the zero propagation and processing delay case.

6 Conclusions

We analyzed the throughput performance of a wireless media access protocol, taking into account the effect of correlated multipath channel fading, capture, and propagation and processing delays. The paper mainly focused on analyzing the effect and exploitation of the channel correlation. For efficient access on the uplink, the protocol made use of the uplink channel status information, which was conveyed to the mobiles through a busy/idle flag broadcast on the downlink. A first-order Markov model was used to describe the packet success/failure process on the correlated Rayleigh fading channel. A closed-form expression for the throughput was derived by modeling the system as a Markov chain. The analytical results obtained through the first-order Markov approximation of the channel were compared to those obtained from an i.i.d. channel model. The Markovian fading channel model provided better performance results than the i.i.d. channel model. Simulations showed that a first-order Markov approximation of the fading process is quite accurate. An enhanced protocol with an error-detect feature, taking advantage of the channel memory, was proposed and shown to improve performance under slow fading conditions. The protocol with a retransmission feature to recover erroneous data packets was found to perform best under fast fading conditions. Further, a parameterized retransmission strategy which allows a maximum of 2 or 3 retransmissions was shown to result in good performance over the range of mobile speeds of interest. The header packet capture phenomenon was shown to result in about 11% to 18% improvement in system throughput under slow fading conditions. A simplified analysis was carried out to show the effect of non-zero propagation and processing delay on the throughput performance of the access protocols. A propagation and processing delay of one slot duration was shown to result in about 13% to 20% degradation in maximum throughput compared to zero propagation and processing delay. Finally, it is noted that priority and fairness are important issues in access protocols. Extensions of the current analysis to address the priority and fairness issues are left for future investigation.

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Figures

Figure 1: Throughput, η, vs message arrival rate, λ. $N = 10$. $g_d = 0.1$. $f_D T = 0.02$. No capture. Instantaneous and error-free feedback.

Figure 2: Maximum throughput, η_{max}, vs normalized Doppler bandwidth, $f_D T$. $N = 10$. $g_d = 0.1$. No capture. $\lambda = 1/N$. Instantaneous and error-free feedback.

Figure 3: Maximum throughput, η_{max}, vs uplink fading margin, F, in dB. N = 10. $g_d = 0.1$. No capture. $\lambda = 1/N$. $f_D T = 0.02$. Instantaneous and error-free feedback.

Figure 4: Maximum throughput, η_{max}, vs average message length, $1/g_d$. $N = 10$. $f_D T = 0.02$. No capture. $\lambda = 1/N$. Instantaneous and error-free feedback.

Figure 5: Maximum throughput, η_{max}, vs capture threshold, B, in dB. N = 10. $g_d = 0.1$. $\lambda = (1 + B)/BN$. $f_D T = 0.02$. Instantaneous and error-free feedback.

Figure 6: Maximum throughput, η_{max}, vs normalized Doppler bandwidth, $f_D T$. Parameterized retransmission ($n_r = 0, 1, 2, 5, 10, \infty$). N = 10. $g_d = 0.1$. $\lambda = 1/N$. No capture. Instantaneous and error-free feedback.

Figure 7: Throughput, η, vs message arrival rate, λ. N = 10. $g_d = 0.1$. $f_D T = 0.02$. No capture. Error-free feedback. One slot propagation and processing delay.

Figure 8: Throughput, η, vs message arrival rate, λ. N = 10. $g_d = 0.1$. $f_D T = 0.02$. No capture. Error-free feedback. Zero and one slot propagation and processing delay.

Figure 9: Throughput, η, vs message arrival rate, λ. N = 10. $g_d = 0.1$. $f_D T = 0.64$. No capture. Error-free feedback. Zero and one slot propagation and processing delay.

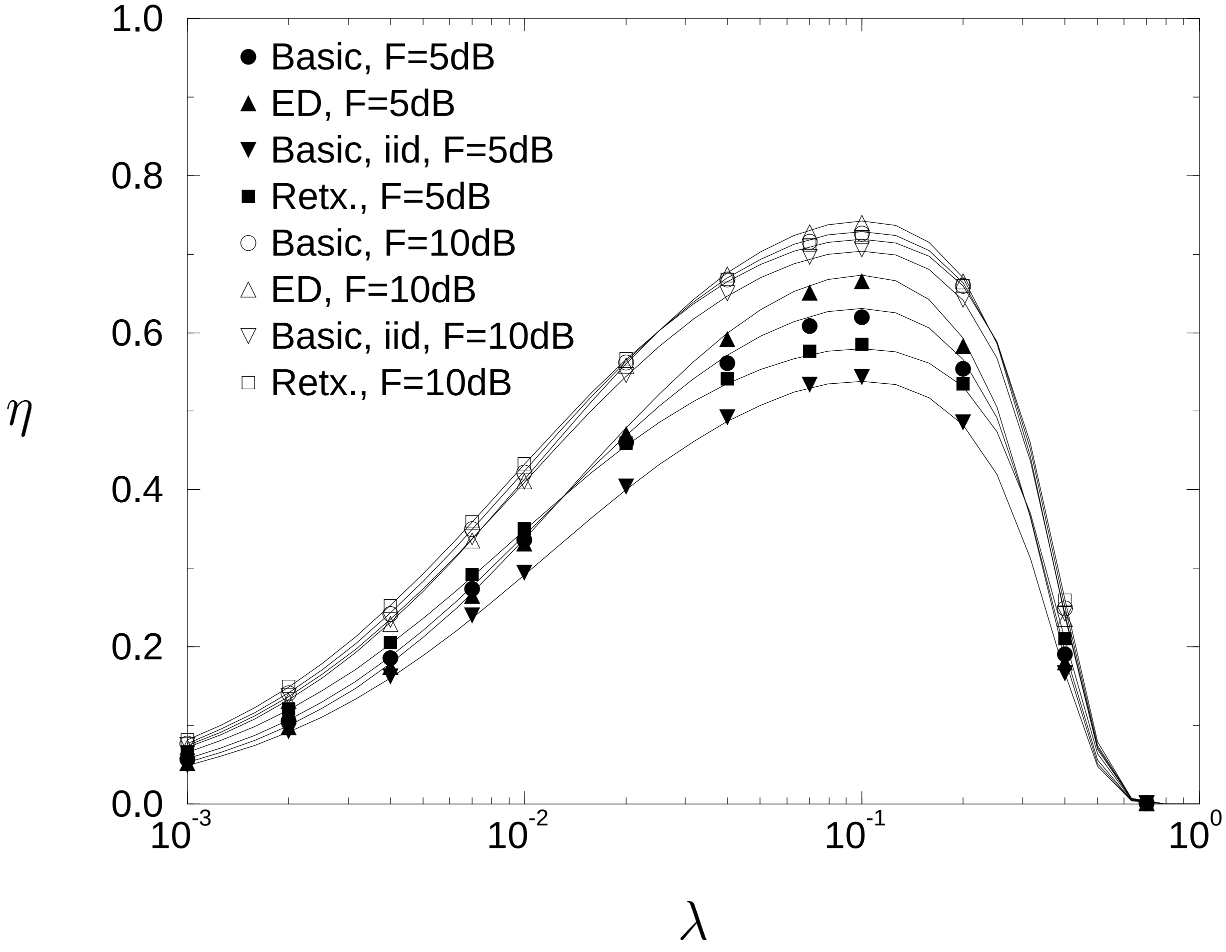

Figure 1: Throughput, η, vs message arrival rate, λ. $N = 10$. $g_d = 0.1$. $f_D T = 0.02$. No capture. Instantaneous and error-free feedback.

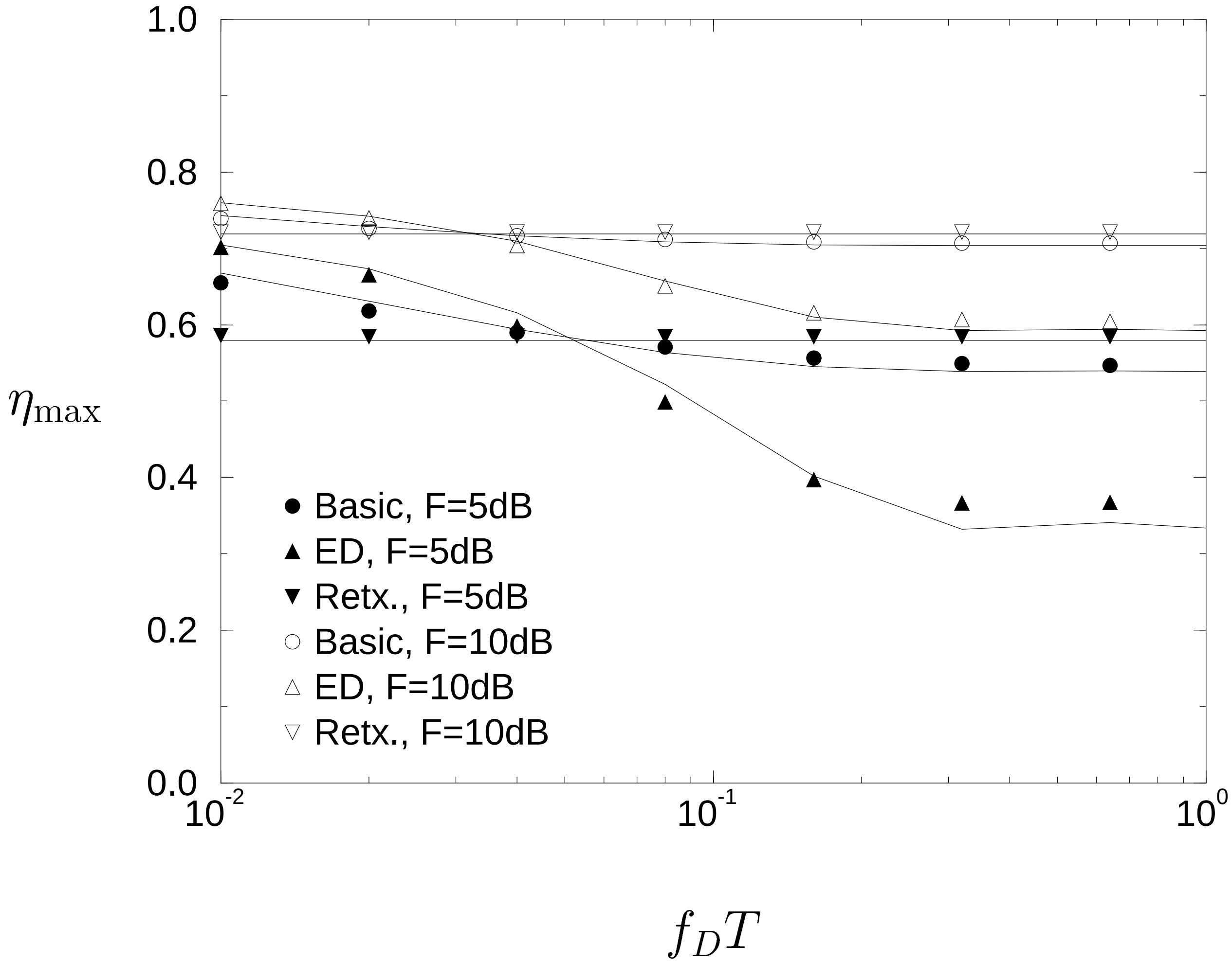

Figure 2: Maximum throughput, η_{max}, vs normalized Doppler bandwidth, $f_D T$. $N = 10$. $g_d = 0.1$. No capture. $\lambda = 1/N$. Instantaneous and error-free feedback.

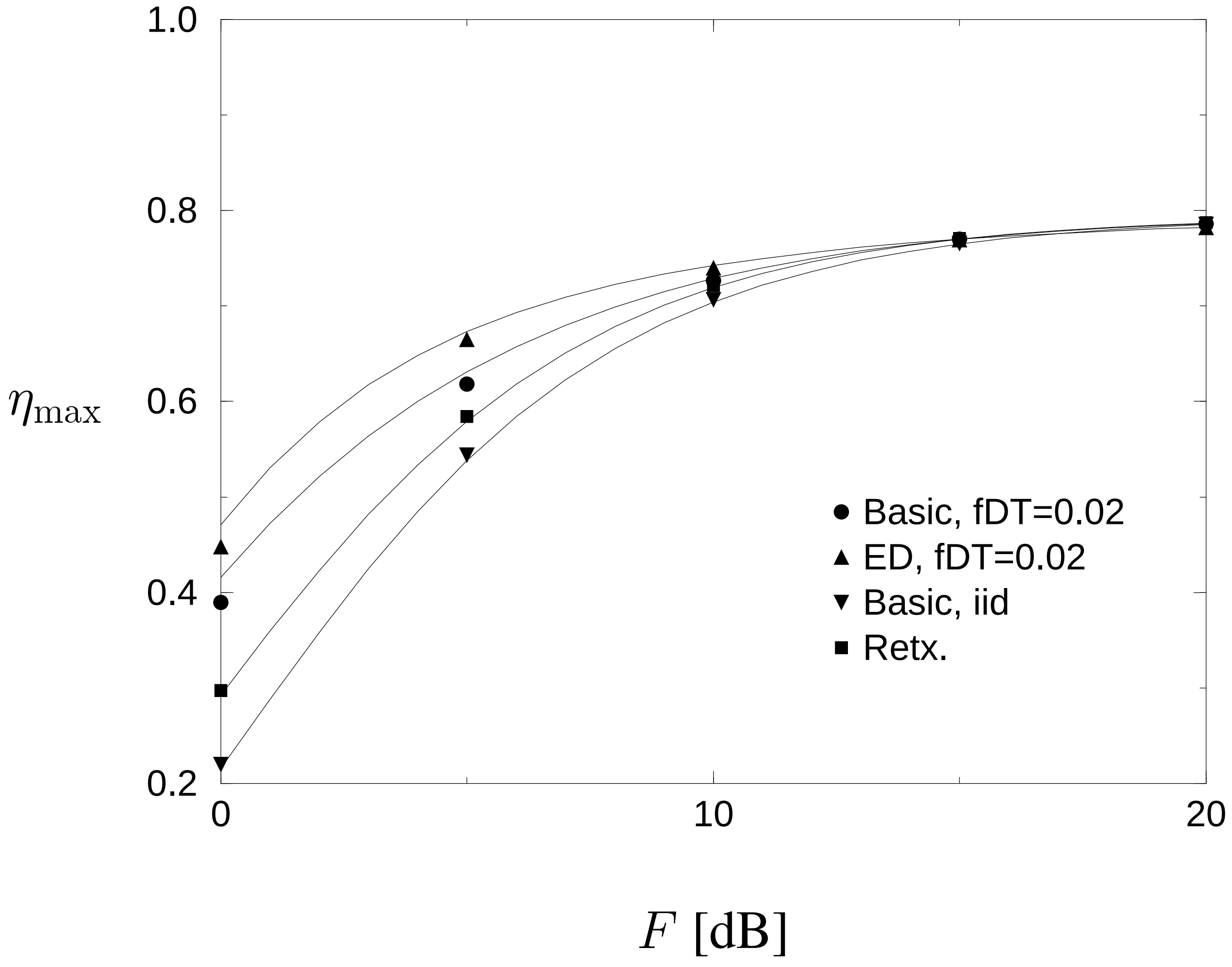

Figure 3: Maximum throughput, η_{max}, vs uplink fading margin, F, in dB. N = 10. $g_d = 0.1$. No capture. $\lambda = 1/N$. $f_D T = 0.02$. Instantaneous and error-free feedback.

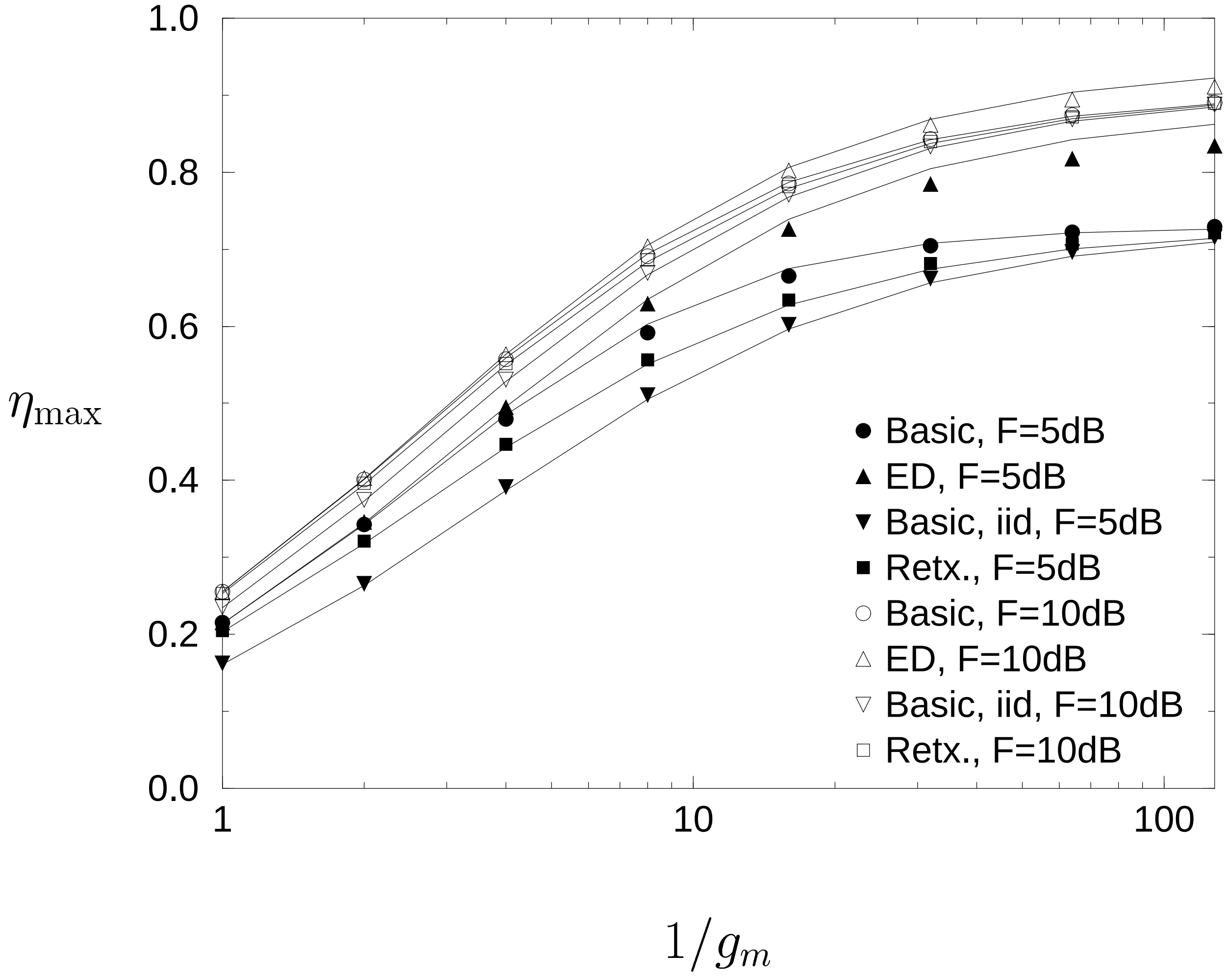

Figure 4: Maximum throughput, η_{max}, vs average message length, $1/g_d$. $N = 10$. $f_D T = 0.02$. No capture. $\lambda = 1/N$. Instantaneous and error-free feedback.

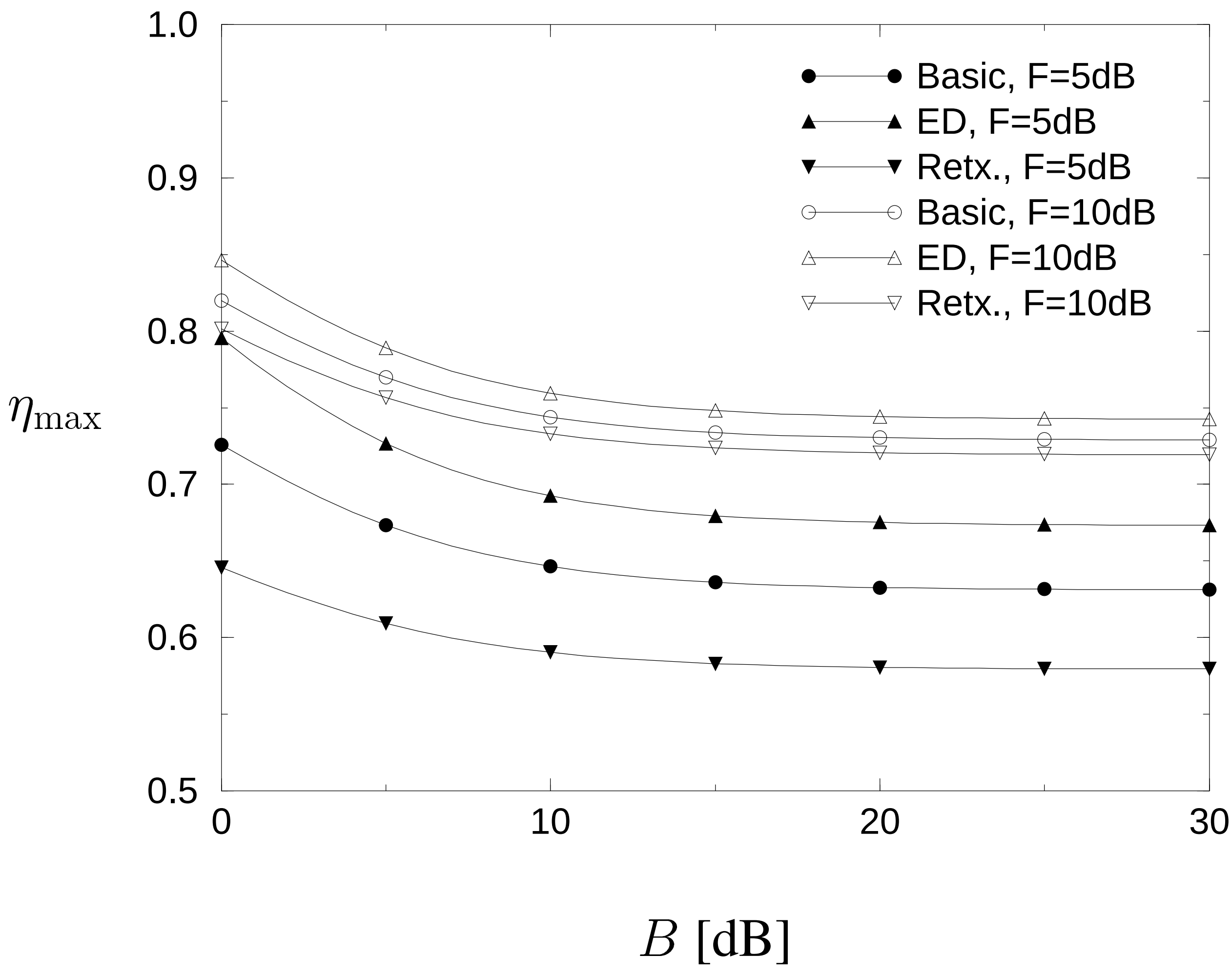

Figure 5: Maximum throughput, η_{max}, vs capture threshold, B, in dB. N = 10. $g_d = 0.1$. $\lambda = (1 + B)/BN$. $f_D T = 0.02$. Instantaneous and error-free feedback.

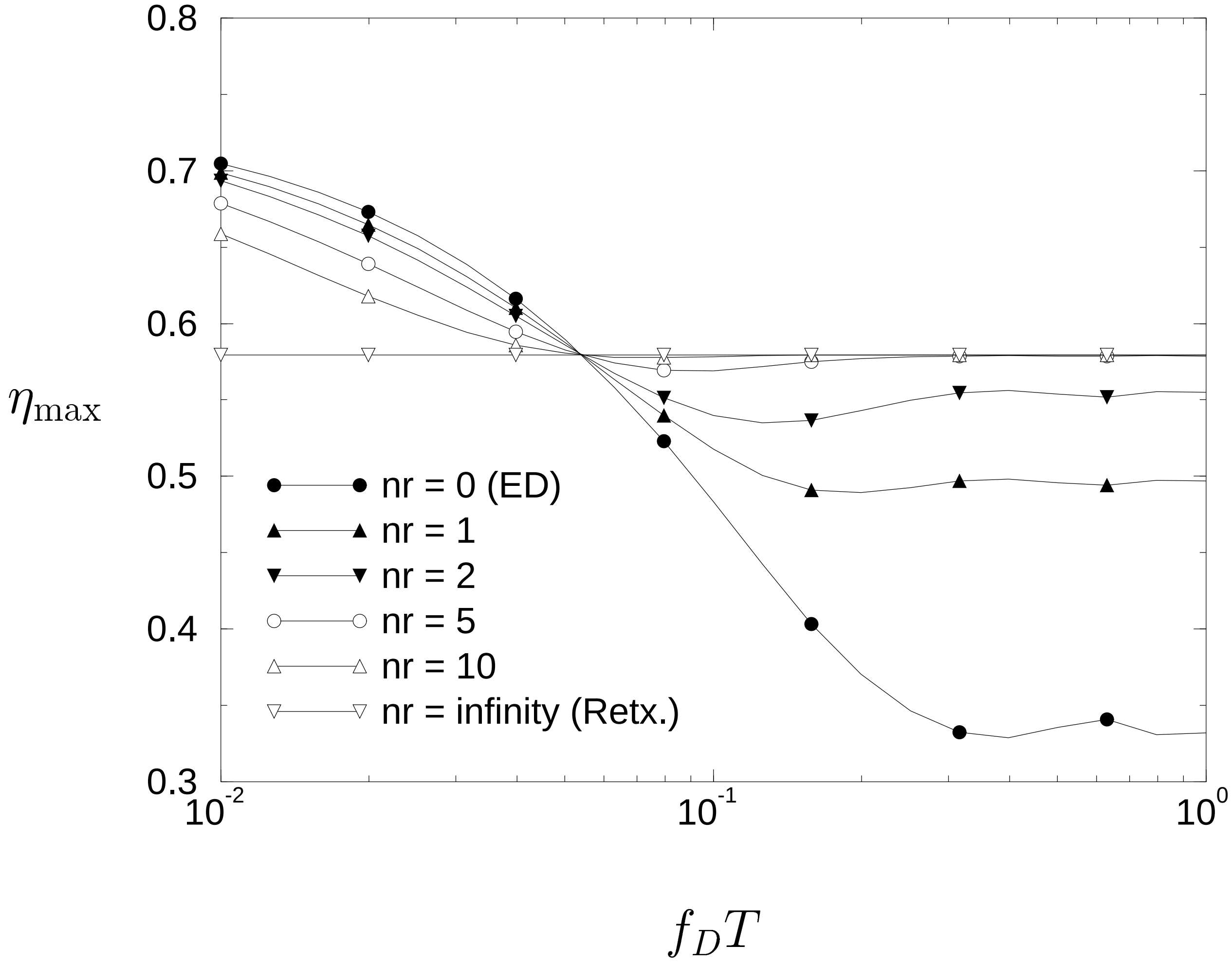

Figure 6: Maximum throughput, η_{max}, vs normalized Doppler bandwidth, $f_D T$. Parameterized retransmission ($n_r = 0, 1, 2, 5, 10, \infty$). N = 10. $g_d = 0.1$. $\lambda = 1/N$. No capture. Instantaneous and error-free feedback.

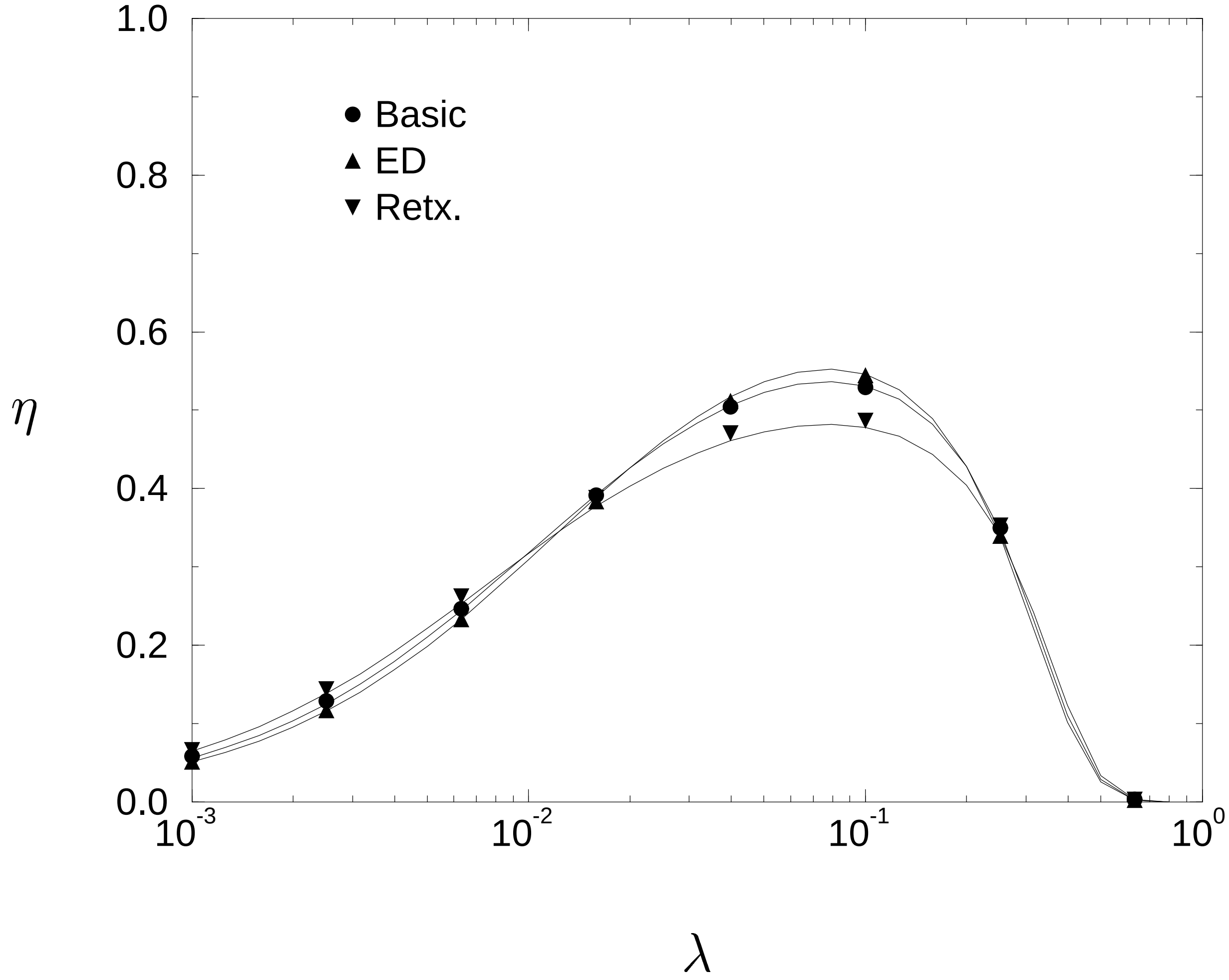

Figure 7: Throughput, η, vs message arrival rate, λ. N = 10. $g_d = 0.1$. $f_D T = 0.02$. No capture. Error-free feedback. One slot propagation and processing delay.

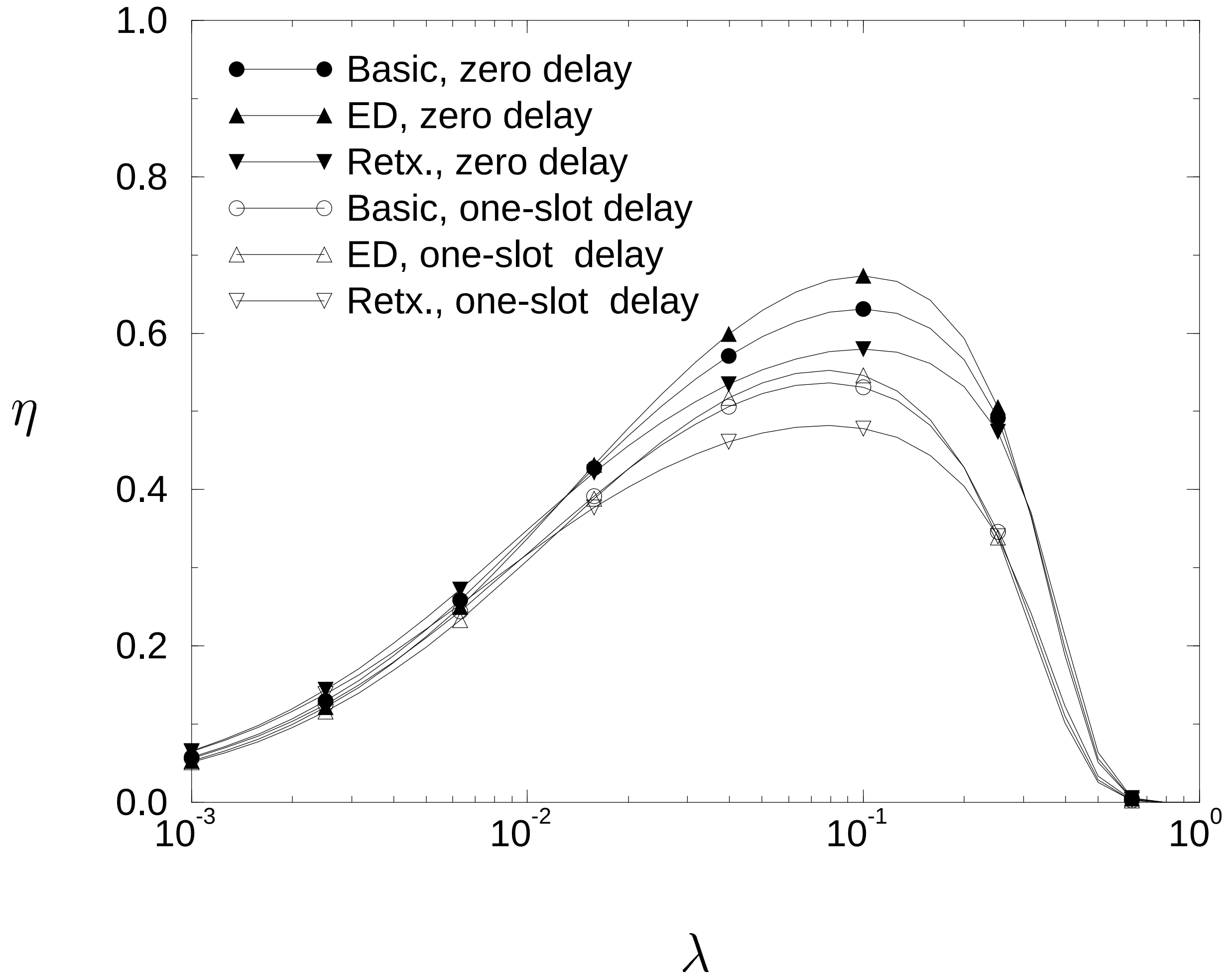

Figure 8: Throughput, η, vs message arrival rate, λ. N = 10. $g_d = 0.1$. $f_D T = 0.02$. No capture. Error-free feedback. Zero and one slot propagation and processing delay.

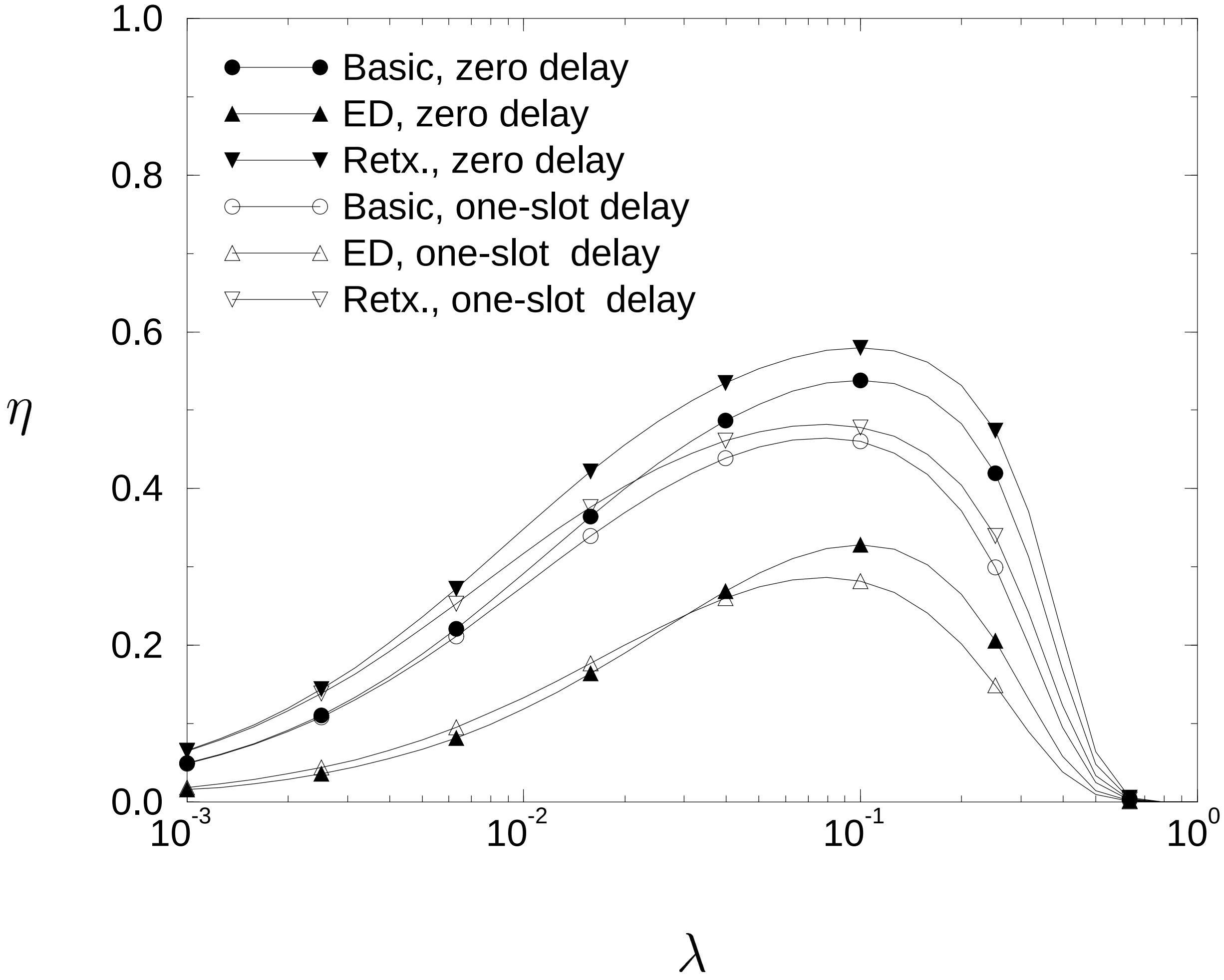

Figure 9: Throughput, η, vs message arrival rate, λ. N = 10. $g_d = 0.1$. $f_D T = 0.64$. No capture. Error-free feedback. Zero and one slot propagation and processing delay.