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STUDIES ON THE PERFORMANCE OF SOME ARQ SCHEMES

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Abstract

This thesis consists of a summary part and seven published articles. All the articles are about performance analysis of ARQ schemes.

Two of the publications study the performance of an ARQ scheme with packet combining, called the EARQ (extended ARQ) scheme. In the packet combining algorithm, the bitwise modulo-2 sum of two erroneous copies of a packet is computed to locate the errors. The packet combining algorithm involves a straightforward search procedure, the computational complexity of which easily becomes prohibitive. As a solution to this, a modified scheme is proposed, where the search procedure is attempted only when there are at most $N_{\max}$ 1s at the output of the modulo-2 adder. In one article, time diversity was utilized, whereas space diversity reception was considered in the other work.

The remaining five publications study the throughput performance of adaptive selective-repeat and go-back-N ARQ schemes, where the switching between the transmission modes is done based on the simple algorithm proposed by Y.-D. Yao in 1995. In this method, α contiguous NACKs or β contiguous ACKs indicate changes from ‘good’ to ‘bad’ or from ‘bad’ to ‘good’ channel conditions, respectively. The numbers α and β are the two design parameters of the adaptive scheme. The time-varying forward channel is modelled by two-state Markov chains, known as Gilbert-Elliott channel models. The states are characterized by bit error rates, packet error rates or fading parameters. The performance of the adaptive ARQ scheme is measured by its average throughput over all states of the system model, which is a Markov chain. A useful upper bound for the achievable average throughput is provided by the performance of an (assumed) ideal adaptive scheme which is always in the ‘correct’ transmission mode. The optimization of α and β is done based on minimizing the mean-square distance between the actual and the ideal performance curves. Methods of optimizing the packet size(s) used in the adaptive selective-repeat scheme are also proposed.

Keywords: adaptive protocol, automatic repeat request, diversity combining, error control, Markov model, packet combining

Tiivistelmä

Tämä väitöskirja sisältää johdanto-osan ja seitsemän julkaistua artikkelia. Kaikki artikkelit tutkivat ARQ-protokollien suorituskyvyn analysointia.

Julkaisuista kaksi käsittelee EARQ:ksi nimettyä laajennettua ARQ-protokollaa, missä pyritään löytämään vastaanotetuissa paketeissa olevia virheitä yhdistämällä saman datapaketin kaksi virheellisinä vastaanotettua kopiota keskenään. Yhdistäminen tapahtuu laskemalla bittivektorit yhteen modulo 2. Algoritmiin kuuluu suoraviivainen oikean koodisanan hakuprosessi, jonka laskennallinen kompleksisuus kasvaa nopeasti kohtuuttoman suureksi. Tämän ongelman ratkaisuksi ehdotetaan muunnettua algoritmia, missä hakurutiini käynnistetään vain, jos modulo-2-summaimen ulostulossa on korkeintaan $N_{\max}$ ykköstä. Artikkeleista toisessa käsitellään aika- ja toisessa puolestaan paikkadiversiteetin hyödyntämistä.

Loput viisi julkaisua tutkivat adaptiivisten ARQ-protokollien suorituskykyä. Tehokkuutta mitataan läpäisyllä (throughput), ja käsiteltävänä ovat sekä valikoivaa toistoa hyödyntävät SR-protokollat (selective repeat) että liukuvaa lähetysikkunaa käyttävät go-back-N-protokollat. Tarkasteltavissa adaptiivisissa protokollissa lähettimellä on kaksi tilaa, joiden välillä siirrytään Y.-D. Yaon vuonna 1995 esittelemän algoritmin mukaisesti: α peräkkäistä negatiivista kuittausta (eli uudelleenlähetyspyyntöä) merkitsee, että kanavan tila on todennäköisesti muuttunut 'hyvästä' 'huonoksi'; β peräkkäistä positiivista kuittausta puolestaan tulkitaan merkiksi päinvastaisesta muutoksesta. Luvut α ja β ovat tällä periaatteella toimivan adaptiivisen protokollan kaksi suunnitteluparametria. Aikavaihtelevia lähetyskanavia mallinnetaan kaksitilaisilla Markovin ketjuilla, joita nimitetään Gilbert-Elliott-kanavamalleiksi. Kanavan tila määritellään joko bittivirhetodennäköisyyden, pakettivirhetodennäköisyyden tai häipymisparametrien avulla. Koko systeemin tilamalli osoittautuu Markovin ketjuksi, ja adaptiivisen protokollan suorituskykyä mitataan laskemalla sen keskimääräinen läpäisy yli kaikkien tilojen. Saavutettavissa olevalle keskimääräiselle läpäisylle saadaan yläraja laskemalla läpäisy hypoteettiselle ideaaliselle adaptiiviselle protokollalle. Optimaalinen (α, β)-pari on se, jota vastaava läpäisykäyrä on lähimpänä ideaalista käyrää pienimmän neliövirheen mielessä. Lisäksi julkaisuissa on tutkittu käytettävän pakettikoon optimointia adaptiivisille SR-protokollille.

Avainsanat: adaptiivinen protokolla, ARQ, diversiteettiyhdistely, Markov-malli, virheentarkkailu

Preface

The research work for this thesis was done at the Communications Laboratory of Helsinki University of Technology during 1997–2005. In the early years, the work was funded by the Academy of Finland, the National Technology Agency of Finland (TEKES), Nokia Research Center (NRC) and Ericsson Ltd. Since the autumn of 2002, my work has been financed by the Graduate School in Electronics, Telecommunications and Automation (GETA), and the Academy of Finland under grant 100500. The additional funding from Jenny and Antti Wihuri Foundation is also gratefully acknowledged.

I am grateful to Professor Patric Östergård, the supervisor of this thesis, for helping me to get this work finished and for many helpful comments.

I thank Dr. Shyam Chakraborty for directing the research work and for several years of interesting collaboration.

I appreciate very much the thorough job done by the two pre-examiners of my thesis, Prof. T. Aaron Gulliver and Dr. Ulrich Tamm. I received several useful comments, which improved the quality of the manuscript.

I also want to thank the entire staff of the Communications Laboratory for making it such a pleasant working environment, as well as all the other people who have one way or another helped me.

Espoo, 9th March, 2006

Markku Liinaharja

Contents

List of Abbreviations

ACK	positive acknowledgement
ARQ	automatic-repeat-request
AWGN	additive white gaussian noise
BER	bit error rate
CRC	cyclic redundancy check
EARQ	extended ARQ
GBN	go-back-N
G-E	Gilbert-Elliott
HARQ	hybrid ARQ
HARQ-I	type I HARQ
HARQ-II	type II HARQ
HMM	hidden Markov model
ISR	ideal selective-repeat
NACK	negative acknowledgement
OBI	observation interval
PER	packet error rate
RCC	rate compatible convolutional
SW	stop-and-wait
SR	selective-repeat
XOR	exclusive-OR

List of Symbols

L	transmission mode for low error rates
H	transmission mode for high error rates
α	OBI length in mode L
β	OBI length in mode H
η	throughput
T	packet throughput
N	transmitter buffer length in GBN protocol
n	packet size in bits
ϵ	bit error rate
h	number of overhead bits per packet
P_c	probability of error-free transmission
n_{opt}	optimal packet size
ϵ_{co}	crossover BER
P_f	probability of acknowledgement error
P_e	packet error rate
m	number of copies used in multicopy GBN scheme
P_{co}	crossover PER
G	good state of forward channel
B	bad state of forward channel
γ	transition probability from G to B
δ	transition probability from B to G
g	good state of return channel
b	bad state of return channel
λ	transition probability from g to b
μ	transition probability from b to g
η_{ave}	average throughput of adaptive SR scheme
η_{ideal}	average throughput of ideal adaptive SR scheme
π_G	steady-state probability of state G
π_B	steady-state probability of state B
ϵ_1	BER in state G
ϵ_2	BER in state B
$P_{\text{f,ave}}$	average probability of acknowledgement error
T_{ave}	average packet throughput of adaptive GBN scheme
T_{ideal}	average packet throughput of ideal adaptive GBN scheme
$P_{e,1}$	PER in state G
$P_{e,2}$	PER in state B

List of Publications

[**P1**] S.S. Chakraborty, E. Yli-Juuti, and M. Liinaharja. An ARQ scheme with packet combining. *IEEE Communications Letters*, **2**:200–202, July 1998.

[**P2**] S.S. Chakraborty and M. Liinaharja. Analysis of adaptive SR ARQ scheme in time-varying channels. *Electronics Letters*, **36**:2036–2037, November 2000.

[**P3**] S.S. Chakraborty and M. Liinaharja. Performance analysis of an adaptive SR ARQ scheme for time-varying Rayleigh fading channels. In *Proceedings of ICC 2001*, pages 2478–2482, June 2001.

[**P4**] M. Liinaharja and S.S. Chakraborty. Analysis and optimization of an adaptive selective-repeat scheme for time-varying channels with feedback errors. *AEÜ International Journal of Electronics and Communications*, **56**:177–186, March 2002.

[**P5**] S.S. Chakraborty, M. Liinaharja, and P. Lindroos. Analysis of an adaptive selective-reject scheme in time-varying channel with non-negligible round-trip delay and erroneous feedback. *Wireless Personal Communications*, **26**:347–363, September 2003.

[**P6**] S.S. Chakraborty, M. Liinaharja, and P. Lindroos. Analysis of adaptive GBN schemes in a Gilbert-Elliott channel and optimization of system parameters. *Computer Networks*, **48**:683–695, July 2005.

[**P7**] S.S. Chakraborty, M. Liinaharja, and K. Ruttik. Diversity and packet combining in Rayleigh fading channels. *IEE Proceedings - Communications*, **152**:353–356, June 2005.

For all the publications in this thesis, the work of the research group has been led by Dr. Chakraborty in the sense that he has outlined the topic and scope of each article. The author of this thesis has been responsible for most of the mathematical analysis and computing in these publications. The contributions of the author(s) are described in more detail below.

In publication [P1], the author of this thesis derived Equation (4), had an important part in writing Sections II–IV together with the other authors, and also wrote and ran the computer programs both for simulation of the scheme and for the numerical computation of the approximate throughput. The idea of the ARQ scheme that was studied came from the first author, who also wrote most of the text.

In publication [P2], the author of this thesis wrote the second and the third section, and did all the analysis and numerical computations. The observation that neither Yao nor Annamalai & Bhargava in their papers cited here had used a time-varying channel model was made by the author of this thesis. The formulation of the Markov model of the system and the definition of the ideal performance curve as an upper bound were also due to the author of this thesis. These findings played an important part also in [P3]–[P6]. The first author wrote the introduction, and the ideas of using half-size packets and acknowledging them in pairs are also due to him.

Publications [P3] and [P4] were written and the contents were produced almost completely by the author of this thesis. In [P4], this author's contributions include the formulation of the optimization problem of the parameters α and β in Yao's algorithm for an adaptive ARQ scheme operating on a two-state Markov channel (a similar approach was used also in [P5] and [P6]) and, in particular, a new method for the optimization of the packet lengths used by the adaptive SR scheme (this method was used also in [P5]). Dr. Chakraborty contributed Fig. 2 in [P3], and some improving comments in both of these publications.

In publications [P5] and [P6], the introductions and also most of the other parts of the text were written by the first author. Section 3.3 in [P6] was written completely by the author of this thesis. The Markov models of the systems were constructed by the author of this thesis and Mr. Lindroos. Mr. Lindroos contributed the idea of how to make the number of states in the system model independent of the round-trip delay; all the rest of the performance analysis of the adaptive schemes and all the numerical computations were done by the author of this thesis alone.

In publication [P7], the author of this thesis assisted in writing the literature survey in the introduction, did the brief analysis in Section 3, plus wrote and ran the simulation programs. The paper was mostly written by the first author. All the simulation programs were built on top of the Matlab implementation of the Jakes channel model provided by Mr. Ruttik, who also wrote Section 4.1.

Chapter 1

Introduction

1.1 Motivation

During the past couple of decades, digital communication has become ubiquitous. The simplest digital communication system, a communication *link*, is shown in its most abstracted form in Figure 1.1. The task of *the transmitter* is to render the message suitable for transmission over *the channel*. The channel is the physical medium that is used to convey the information from the transmitter to the *receiver*. In [30, p. 15], communication channels are divided into two basic groups: channels based on *guided propagation* (telephone channels, coaxial cables and optical fibres), and channels based on *free propagation* (wireless broadcast channels, mobile radio channels and satellite channels). The receiver reconstructs the message from the received signal.

Figure 1.1: Digital communication link

The communication channels exhibit many kinds of non-ideal behaviour, such as additive noise, fading caused by multipath propagation, and intersymbol interference. As a result from these phenomena, the received signal is often so badly distorted that the message cannot be reconstructed unless some kind of *error control* is used.

1.2 Background: ARQ and FEC

There are two basic approaches to error control in digital communications: *forward error correction (FEC)* and *automatic repeat request (ARQ)* [44, 77].

In the FEC systems, parity-check bits are added to each transmitted message block to form a codeword based on the error-correcting code that is being used. The receiver attempts to locate and correct the errors that it has detected in a received word. After the error-correction procedure, the decoded data block is delivered to the end user. A *decoding error* occurs if the output of the decoder is a different codeword than the one that was originally transmitted. The FEC systems are designed for use in simplex channels, where information flows in only one direction.

In an ARQ scheme, a high-rate error-detecting code is used together with some retransmission protocol. If the receiver detects errors in the received word,it generates a retransmission request, or a negative acknowledgement (NACK). If no errors are detected in the received word, the receiver sends a positive acknowledgements, called an ACK, to the transmitter. The most widely used error-detecting codes are the cyclic redundancy check (CRC) codes because of the ease of implementation. Unlike the FEC systems, the ARQ schemes require the presence of a *feedback channel*.

The *stop-and-wait (SW) scheme* is the simplest of all ARQ schemes. In this scheme, the transmitter sends a codeword to the receiver and waits for an acknowledgement. If an ACK comes, the transmitter sends the next codeword in the queue; in case of a NACK, the same codeword is retransmitted, and this process continues until the codeword is accepted. If the system has a significant round-trip delay, the SW scheme becomes quickly very inefficient because of the idle time that the transmitter spends waiting for acknowledgements.

In the *go-back-N (GBN)* scheme, the transmitter sends codewords continuously and stores them to wait for acknowledgements; buffer space for N packets is needed at the transmitter. The acknowledgement for a codeword arrives after a round-trip delay, during which $N-1$ other codewords are transmitted. When a NACK is received for codeword i, the transmitter stops transmitting new codewords, goes back to codeword i and retransmits it and the $N-1$ following codewords. The receiver discards the erroneously received codeword i and all $N-1$ subsequently received words, regardless if they are error-free or not.

Another continuous ARQ strategy, *selective-repeat (SR)* ARQ, is much more efficient than GBN, since only negatively acknowledged codewords are retransmitted. After resending a negatively acknowledged codeword, the transmitter continues transmitting new codewords in the transmitter buffer. Whereas the GBN scheme automatically preserves the original order of the codewords, the receiver in the SR scheme must have some buffer space to store the correctly received codewords that can not yet be released.

If some error-correcting capability is added to an ARQ scheme, we have a *hybrid ARQ (HARQ)* scheme. HARQ schemes, which are thus combinations of ARQ and FEC, are discussed in Section 2.5.

1.3 Scope and Structure of the Thesis

This thesis consists of mainly theoretical studies on the performance of some ARQ schemes. Two kinds of schemes have been studied: (i) ARQ schemes with diversity combining in [P1] and [P7], (ii) adaptive ARQ schemes in [P2]–[P6].

An ARQ scheme with packet combining (using time diversity) is studied in a random-error channel environment in [P1]. Approximate performance analysis is presented and the results are compared to simulations. The packet combining algorithm, based on computing the bit-wise modulo-2 sum of two erroneous copies of a packet was originally proposed by Sindhu in [67]. In [P7], the same scheme, referred to as the EARQ scheme (for 'extended ARQ') is used in a wireless communication system where there are two antennas at the receiver. That is, spatial diversity is utilized. The performances of the EARQ scheme and three other schemes are compared in fading channels by simulations. A sum-of-sinusoids model, derived from that proposed by Jakes in [32], is used to represent Rayleigh fading channels. The sinusoids have random frequency inside the Doppler spectrum, and random phase.

In [79], Yao proposes an adaptive GBN scheme with two transmission modes, denoted by L and H and meant for 'good' and 'bad' channel conditions, respectively. What is significant for this thesis is that he suggests a simple algorithm for detecting channel state changes, which is based on observing the acknowledgments and is defined by two integer-valued parameters: α and β: if α contiguous NACKs are received by the transmitter when it is in mode L, it is concluded that the channel conditions are deteriorating and the transmitter switches to mode H; if β ACKs are received contiguously while the transmitter is in mode H, mode L is resumed. In [3, 4] and a few other articles, Annamalai and Bhargava study adaptive GBN and SR schemes based on Yao's channel sensing algorithm; they also attempt to optimize the design parameters α and β. However, no time-varying channel model where the channel state actually changes has been specified in these articles, and the same is true for Yao's original paper. This is the starting point for the publications [P2]–[P6] of this thesis. In all these papers, the performance of the adaptive SR or GBN scheme using Yao's algorithm is evaluated in two-state Markovian channel environments under varying assumptions about the return channel and the round-trip delay.

The summary of the thesis is organized as follows. Chapter 2 reviews some basic concepts, including some channel models, performance measures of ARQ schemes and hybrid ARQ. Chapter 3 is a survey of the contents of publications [P2]–[P6], while Chapter 4 covers publications [P1] and [P7]. The summaries of all seven publications are provided in Chapter 5, and finally some concluding remarks are made in Chapter 6.

Chapter 2

Basic Concepts

2.1 Channel Models

In order to evaluate the performance of error control systems, we must model the circumstances where they operate. Here we take a somewhat limited perspective on modeling communication channels, and by a *channel model* we mean a mathematical model for the noise process or the error process associated with the communication channel.

2.1.1 Discrete-Time Models

Most of the channel models considered in this work are *discrete-time* models, which are characterized by the values of bit error rate (BER) or packet error rate (PER). Depending on whether the time unit of the model is the transmission time of one bit or one packet, these models can be divided into *bit-level* and *packet-level* models.

The simplest discrete-time model is the memoryless binary symmetric channel (BSC) [27, 59], which is also often referred to as the *random-error* channel. In a BSC, a bit is received in error with a certain probability (the BER), independently of all the other bits. As a result, the number of bit errors in a received n-bit packet is binomially distributed; if the BER is equal to ϵ, the PER is given by

$$P_e(n, \epsilon) = 1 - (1 - \epsilon)^n. \tag{2.1}$$

A schematic picture of the BSC where the BER is equal to p is shown in Figure 2.1.

In many practical channels, especially in the presence of fading, the bit errors are not statistically independent, but occur in bursts. They are called channels with *memory*, or *burst-error* channels. If FEC is used and the error-correcting code is designed to correct random errors, the channel errors can be made to look more random by using an *interleaver* after encoding the data block and a *de-interleaver* before decoding the received word [59, 68]. On the other hand, ARQ

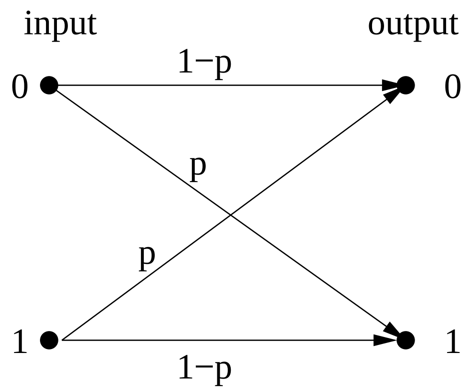

Figure 2.1: The binary symmetric channel

schemes typically perform better with bursty errors than with random errors if the average error probability is the same.

Hidden Markov models (HMMs) have become popular in modelling communication channels with memory [35, 71]. This is because they can be easily fitted to experimental data, and many important statistics can be evaluated in closed form.

In [28], Gilbert proposed a simple HMM to model burst-error channels. This model is a two-state Markov chain, where the probability of a bit error is 0 in state G (for 'good') and of the order of $1/2$ in state B (for 'bad'). This is a bit-level model where a state transition (possibly back to the same state) occurs after each transmitted bit. Elliott generalized this model in [25] by allowing a small non-zero bit-error probability in state G. This kind of HMMs are consequently known as Gilbert-Elliott (G-E) models.

The HMMs can also be used to model random-error channels where the error rate varies with time. This kind of two-state HMM was used for example in [48] and has also been assumed in [P2]. Figure 2.2 shows the state transition diagram of this packet-level model, in which the states G and B are characterized by their BER values, ϵ_1 and ϵ_2 ($\epsilon_1 \ll \epsilon_2$). The transition probabilities from G to B and from B to G are denoted by γ and δ, respectively, and the time interval between state transitions is assumed to be equal to the transmission time of one packet. The occupation times of states G and B are geometric random variables with means $1/\gamma$ and $1/\delta$, respectively. The steady-state probabilities of being in states G and B are given by

$$\pi_G = \frac{\delta}{\gamma + \delta}, \tag{2.2}$$

$$\pi_B = \frac{\gamma}{\gamma + \delta}. \tag{2.3}$$

In [P4] and [P5], it was assumed that there is a state transition after each transmitted bit. This was done because it was necessary to make the definition of the channel environment completely independent of the packet size. In this model, the transition probabilities from G to B and from B to G are denoted by γ_0 and δ_0, respectively ($\gamma_0, \delta_0 \ll 1$). Once the packet size is fixed to n bits, an *approximate* packet-level model with parameters $\gamma = n\gamma_0$ and $\delta = n\delta_0$ is adopted.

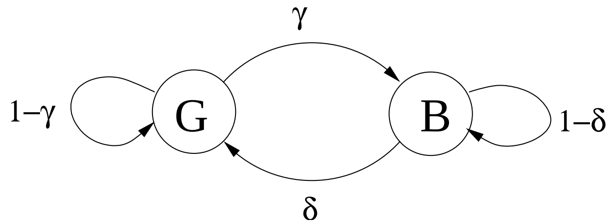

Figure 2.2: The packet-level Gilbert-Elliott channel model

This approach makes it possible to compare the system performance with different packet sizes under approximately similar channel conditions.

A packet-level G-E model was used also in [P6] with the difference that the states G and B where defined by their PER values, $P_{e,1}$ and $P_{e,2}$.

Unreliable return channels have also been considered in the publications of this thesis. It has been assumed that an acknowledgement, ACK or NACK, can be erased, but an ACK cannot become a NACK or vice versa. A similar assumption has been made e.g. in [11, 83]. In [P5] acknowledgements were erased randomly with probability P_f, whereas in [P4] and [P6] a G-E model was assumed also for the feedback channel. In this G-E model, which is assumed to be independent of the forward channel, all the acknowledgements are received successfully in the ‘good’ state, which is denoted by g, but the probability of erasure is P_f in the ‘bad’ state, which is denoted by b.

2.1.2 Threshold Model for Fading Channels

If we use a somewhat simplified description, the average behaviour of a Rayleigh fading channel can be described by two parameters: ρ and f_D. Here, ρ is the ratio between the receiver threshold power and the average received signal power, i.e. a smaller value of ρ means that the channel is better on the average. The threshold power level is selected so that if the instantaneous received signal power is below the threshold value, it can be considered that there is a ‘fade’ going on. The other parameter, f_D, is the maximum Doppler shift, which is equal to v/λ [32], where v is the velocity of the mobile terminal and λ is the carrier wavelength.

By using the level-crossing statistics presented in [32] and making some further simplifying assumptions, DaSilva et al. derived in [21] a simple closed-form expression for the PER in a fading channel described by the threshold model

$$P_e(\rho, f_D, n) = 1 - \exp\left[-\left(\rho + \frac{n f_D \sqrt{2\pi\rho}}{R}\right)\right], \tag{2.4}$$

where n is the packet size in bits and R is the channel transmission rate in bits/s. The following assumptions were made:

- The channel is in one of the two possible conditions at any time: the received signal power is either above (‘non-fade interval’) or below (‘fade interval’) the threshold level.

- A packet is received correctly if and only if the whole packet was contained in a non-fade interval.
- The length of the non-fade interval is exponentially distributed.

This PER expression was used by Annamalai and Bhargava in [4], and in [P3] a two-state packet-level HMM was used, where the states G and B were defined by the parameter combinations $(\rho_1, f_{D,1})$ and $(\rho_2, f_{D,2})$, respectively.

2.2 Performance Measures

2.2.1 Throughput Efficiency

The most important performance measure for the ARQ schemes is the throughput efficiency, or simply the *throughput* η. The throughput is defined as the ratio of the average number of information bits successfully accepted by the receiver per unit time to the total number of bits that could be transmitted per unit time [44]. It can be noted that the throughput of an FEC scheme is a constant irrespective of the channel conditions, and it is equal to the rate of the error-correcting code.

A related performance measure, which we will call the *packet throughput* and denote by T, is defined as the average number of *packets* accepted successfully per one transmission. It is the inverse number of the average number of transmission attempts needed until a packet is received successfully. The difference between η and T is that in computing η, only the information bits are considered 'useful', and hence η represents the 'real' transmission efficiency. The quantities η and T relate to each other as follows:

$$\eta = \frac{k}{n} \cdot T = \frac{(k/n)}{\mathrm{E}[X]}, \tag{2.5}$$

where k/n is the rate of the error-detecting code used by the ARQ scheme, and X is the random variable that represents the number of transmission attempts needed until a packet is received successfully. Naturally, the distribution of X depends on the channel (also the return channel) error statistics.

2.2.2 Other Performance Measures

Besides throughput, many other performance measures of ARQ schemes have been proposed and studied. Most of these are related to the delay in delivering the packets.

As it was pointed out in [37], the total delay of a link layer packet consists of the *transport delay* and the *resequencing delay.* Of these two, the transport delay is further divided into the *queueing delay* and the *transmission delay.* In two early studies [40, 70], the authors derived the generating functions of the transport delay and the transmitter queue length for the basic ARQ schemes when packet errors occur randomly and new packets arrive at the transmitter according to a Poisson process, and in [69] these studies were extended to channels with memory. The mean queue length and mean transport delay for the SR scheme with arbitrary new packet arrival process and random errors were obtained in [2]. In [62], the authors derived the exact distribution of the *delivery delay,* which is the sum of transmission delay and resequencing delay, for the SR scheme, when a two-state Markovian channel model was assumed. Random errors in both forward and return channels were assumed when the mean transmission delay was calculated for the basic ARQ schemes in [46] using signal flow graphs, and the authors generalized the analysis to Markovian channels in [47]. In a recent article [49], Luo et al. calculated the first and second order statistics for the delivery delay of a higher layer packet consisting of a fixed number of link layer blocks, when the SR scheme was used in a random error channel environment. A delay related performance measure, namely the probability that a packet is not delivered within D time slots of its arrival at the transmitter, has been studied in [63, 82].

In the recent years, communications between light portable devices with finite battery resources have become increasingly important. Therefore, *energy efficiency* is also an important performance measure of an error control strategy and has been discussed, e.g., in [18, 81].

2.3 Throughput Performance of Basic ARQ Schemes

The throughput of the SW scheme is given by [44]

$$\eta_{\mathrm{SW}} = \frac{P_c \cdot (k/n)}{1 + DR/n}, \tag{2.6}$$

where P_c is the probability of a successful transmission, D is the *round-trip delay* in seconds, and R is the bit rate of the transmitter. The round-trip delay is an important system parameter, which is defined as the time that elapses after a packet leaves the transmitter before the corresponding acknowledgement arrives [43, p. 459]. Hence, DR/n is the number of packets that could be transmitted during the idle time period when the transmitter is waiting for an acknowledgement.

In the GBN scheme, the transmitter does not stop to wait for acknowledgements after sending a packet, but transmits the next packet in schedule. Both the GBN scheme and the SR scheme are *continuous* ARQ schemes in this sense. If there is a retransmission request for a packet, the transmitter resends N packets, namely the negatively acknowledged one and also all the $N-1$ packets that follow, since the receiver has discarded the previous transmission attempt of those packets. The parameter N, which is the length of the ‘sliding window’ in this protocol, depends on the round-trip delay as follows [10]:

$$N - 1 = \left\lceil \frac{DR}{n} \right\rceil, \tag{2.7}$$

and the throughput of the GBN scheme in a random-error channel is given by [44]

$$\eta_{\mathrm{GBN}} = \frac{P_c \cdot (k/n)}{P_c + (1 - P_c)N} = \frac{(1 - P_e)(k/n)}{1 + (N-1)P_e}, \tag{2.8}$$

where $P_e = 1 - P_c$ is the PER of the channel. Unlike the SW and SR schemes, the throughput of the GBN scheme does not depend only on the average PER, but also on how the packet errors are distributed. It was shown in [42] that if the average PER is the same but errors are burstier, then the throughput of the GBN scheme is higher.

If the receiver of the SR scheme has an infinite buffer (i.e. there cannot be buffer overflow), the throughput is independent of the round-trip delay and is given by [44]

$$\eta_{\mathrm{SR}} = P_c \cdot \left(\frac{k}{n}\right). \tag{2.9}$$

If the round-trip delay is small enough to be negligible, i.e., the acknowledgement for a packet arrives instantaneously after the transmission has ended, then all the three basic ARQ schemes are clearly identical, and we have what is referred to as the *ideal SR (ISR) scheme* [26]. Figure 2.3 shows the throughputs of the three basic ARQ schemes as functions of the BER in a BSC, when $n = 200$, $k = 184$, and $N = 10$ (i.e. the round-trip delay equals the transmission time of 9 packets).

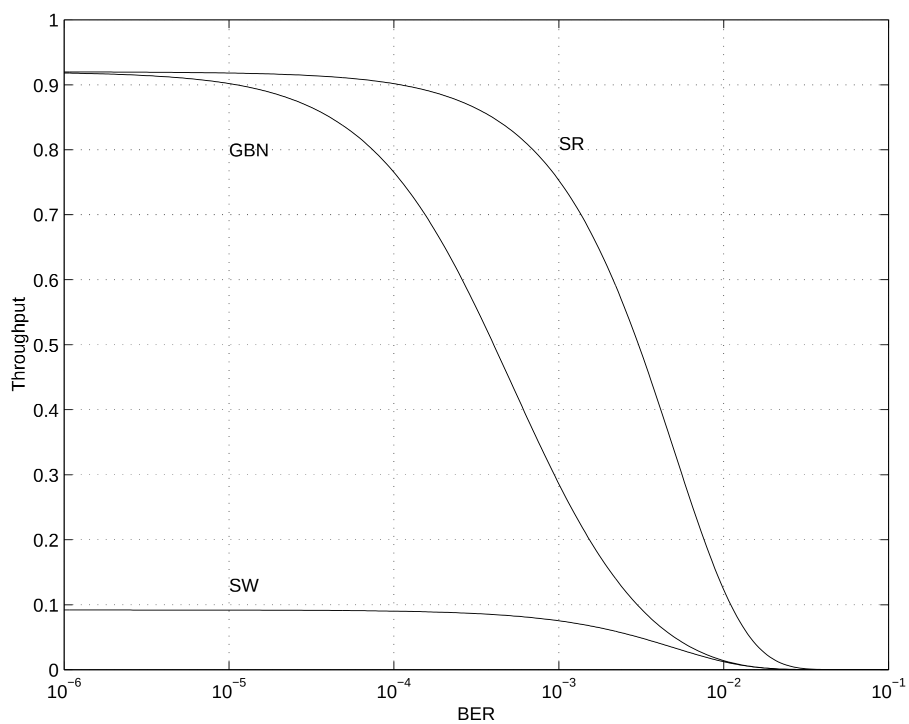

Figure 2.3: A performance comparison of the basic ARQ schemes.

2.4 Improving Basic ARQ Schemes

2.4.1 Selection of the Packet Size

As we have seen, the throughput efficiencies of all the basic ARQ schemes are functions of the packet size n. Consequently, n is a very important design parameter when the performance of the scheme is optimized. There is an obvious trade-off here: for short blocks, the PER is lower and fewer bits are retransmitted in the case of an error; on the other hand, a larger proportion of time is spent on transmitting the actual data bits if longer blocks are used.

Optimization of the packet size has been studied extensively in the past, e.g., [20, 21, 38, 55, 56]. In [38], Kirlin studied the maximization of the quantity that he called the 'transmission efficiency', which was essentially the same as the throughput efficiency η. A different approach was taken in [20], where the author considered random-length messages, which were divided into blocks of fixed length before transmission. The length of these blocks was optimized so that the average 'wasted time' per message was minimized (the time spent on retransmissions, acknowledgements, and transmitting non-data bits was considered 'wasted'). The optimization was performed for all the three basic types of ARQ schemes; both random-error and burst-error channels were considered. In [56], throughput efficiency η was maximized with respect to the packet size for the basic ARQ schemes. An interesting Bayesian approach was proposed in [55], where the expected efficiency of an adaptive ARQ scheme was maximized with respect to the packet size, given the transmission history. Packet size optimization for adaptive ARQ schemes has also been considered in the publications of this thesis, as will be seen later.

The throughput of the SR scheme was given in (2.9). However, in the articles of this thesis, it usually appears in the functional form

$$\eta_{\mathrm{SR}}(n,\epsilon,h) = \frac{n-h}{n}[1-P_e(n,\epsilon)] = \frac{n-h}{n}(1-\epsilon)^n, \tag{2.10}$$

where n is the packet size in bits, ϵ is the channel BER, $P_e(n,\epsilon)$ is the channel PER, and $h = n-k$ is the number of overhead (i.e. non-data) bits per packet. If the acknowledgements get erased on the return channel randomly with probability P_f, the throughput must be multiplied by $(1-P_f)$. If we differentiate η_{SR} with respect to n and set the derivative as zero, and then solve for n, we get

$$n_{\mathrm{opt}} = \frac{h\ln(1-\epsilon) - \sqrt{h^2[\ln(1-\epsilon)]^2 - 4h\ln(1-\epsilon)}}{2\ln(1-\epsilon)}, \tag{2.11}$$

which must naturally be rounded to the one of the adjacent integer values that yields higher throughput.

Figure 2.4 shows η_{SR} as a function of n for BER values 10^{-5}, 10^{-4}, 10^{-3} and 10^{-2}. It can be seen that the optimal packet sizes become smaller and the optima 'steeper' as the BER increases. If $h = 16$, the optimal packet sizes corresponding to BER values 10^{-5}, 10^{-4}, 10^{-3} and 10^{-2} are 1273, 408, 135 and 49 bits, respectively. In Figure 2.5, the SR throughput is shown as a function of the BER for these four packet sizes. We call the value of the BER at which the two curves

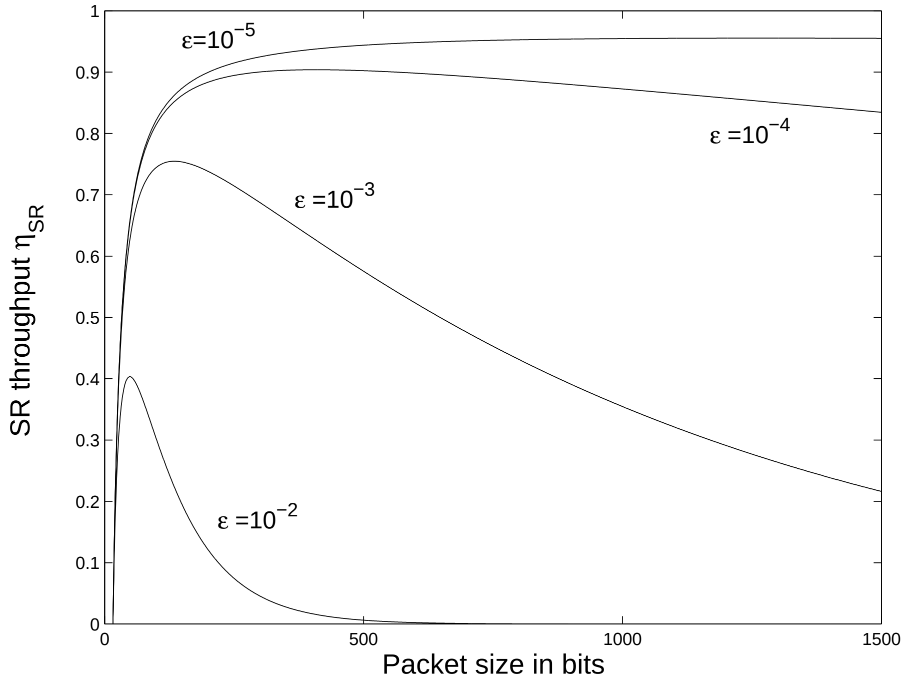

Figure 2.4: The SR throughput as a function of packet size with four different BER values when $h = 16$ and $P_f = 0.02$

corresponding to packet sizes n_1 and n_2 intersect the *crossover BER* of these packet sizes. In particular, if $n_1 = 2n_2$, then the crossover BER is given by [P4]

$$\epsilon_{\mathrm{co}} = 1 - \left[\frac{2(n_2 - h)}{2n_2 - h}\right]^{1/n_2} . \tag{2.12}$$

2.4.2 Use of Multicopy Transmissions

Another potential method of improving the performance of the basic ARQ schemes in poor channel conditions is to use multicopy transmissions, where multiple copies of each data block are sent contiguously before moving on to the next block in schedule. If at least one of the copies is received successfully, the data block is acknowledged positively.

For the SR scheme, the throughput would actually decrease if multicopy transmissions were used because then some successful transmissions would be wasted, which does not happen in the basic SR scheme. However, if the buffer space at the receiver is severely limited, the probability of buffer overflow can be decreased by sending multiple copies of the packet in the retransmissions. A modified SR scheme based on this idea was proposed and analyzed by Weldon in [75]. If the delay performance is considered instead of throughput, it was shown in [80] that in some burst-error channels the mean transmission delay of a packet in the SR scheme can be reduced by sending two identical copies of the packet separated by a fixed delay at each transmission attempt.

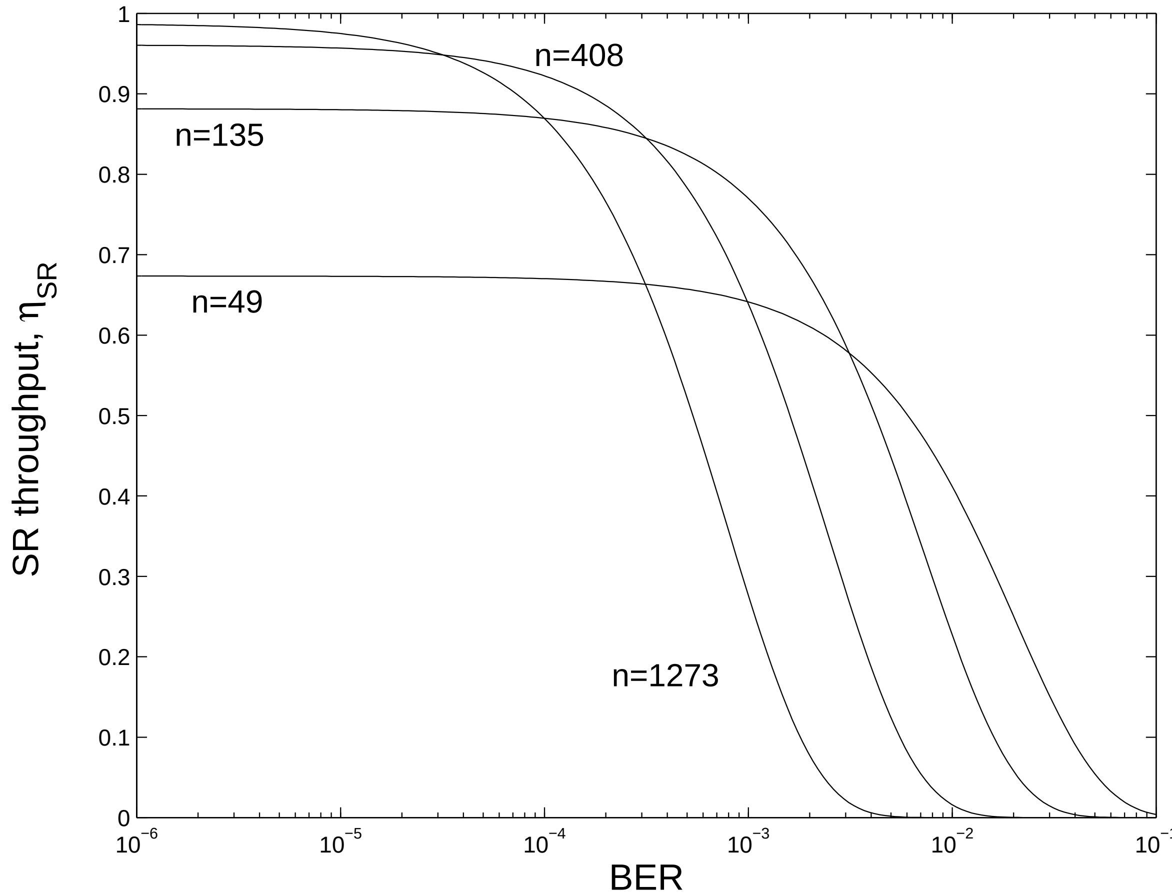

Figure 2.5: The SR throughput as a function of the BER with four different packet sizes when $h = 16$ and $P_f = 0$

In the publications of this thesis, multicopy transmissions are considered with the GBN scheme. An early related work was [64], where Sastry suggested a modified GBN scheme in which, after a retransmission request, the same data block is sent repeatedly until an ACK is received for it. GBN schemes with multicopy transmissions have also been studied e.g., in [9, 10]. The delay performance of multicopy GBN schemes was studied in [22]. In [10], the authors showed that if the GBN ARQ scheme is used in a stationary channel with random packet errors, then the optimal strategy, which maximizes the packet throughput, is to use the same number m of copies in all transmission attempts of a data block. The optimal value of m, however, depends on the PER and the round-trip delay.

If the PER is P_e and feedback errors occur randomly with probability P_f, then the packet throughput of the m-copy GBN scheme is given by [79]

$$T_{\text{GBN}}(N, m, P_e, P_f) = \frac{1-[1-(1-P_e)(1-P_f)]^m}{m+(N-1)[1-(1-P_e)(1-P_f)]^m}\,. \tag{2.13}$$

Figure 2.6 shows T_{GBN} as a function of the PER for $m = 1, 2, 3, 4$ when $N = 10$ and $P_f = 0$. It can be seen that the performance can be improved significantly by selecting the value of m optimally based on channel state information. The PER value at which the curves corresponding to $m = m_1$ and $m = m_2$ intersect is called the *crossover PER*. In the interesting particular case where $m_1 = 1$ and $m_2 = 2$, the crossover PER can be shown to be [P6]

$$P_{\text{co}} = \frac{\frac{1}{N} - P_f}{1 - P_f}\,. \tag{2.14}$$

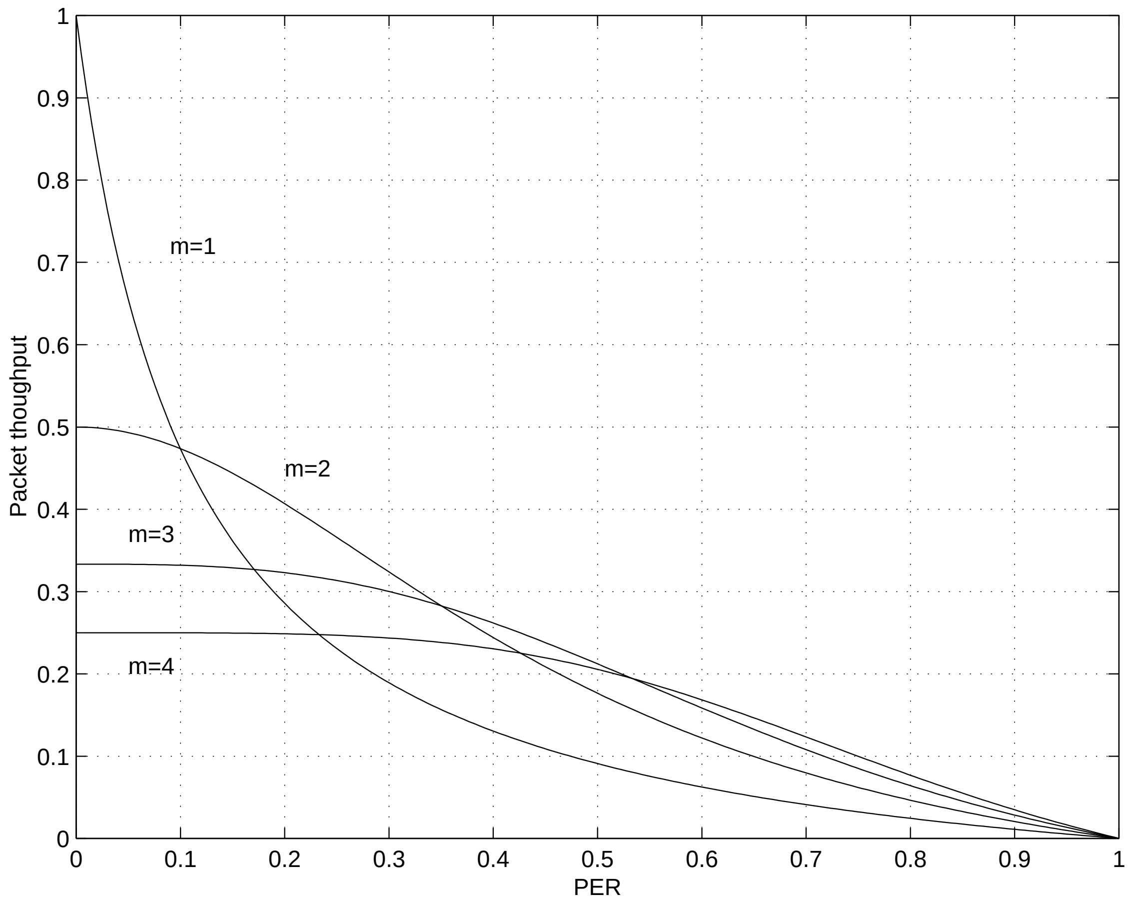

Figure 2.6: Comparison of m-copy GBN schemes with different values of m

2.5 Hybrid ARQ and Packet Combining

The combinations of ARQ schemes and FEC are known as hybrid ARQ (HARQ) schemes [43, 77]. These schemes are further classified into type-I (HARQ-I) and type-II (HARQ-II) schemes.

2.5.1 HARQ-I Schemes

In HARQ-I schemes, all the transmission attempts of a packet are identical codewords containing redundant bits for both error detection and error correction. There are two different ways to accomplish this. In the early HARQ-I schemes, such as those studied in [65, 66], one block code was used simultaneously for error detection and error correction. Another approach is to use two codes: an inner code, which is used for error correction (and possibly for simultaneous error detection if it is a block code), and an outer code, which is used for error detection only. This kind of HARQ-I schemes which use concatenated coding and hence require two encoders at the transmitter and two decoders at the receiver have been proposed and analyzed for example in [8, 24, 36].

Generally speaking, the HARQ-I schemes are best suited for channel environments where the level of noise and interference is fairly constant. Then the error-correcting capability of the FEC part of the scheme can be designed so that most of the erroneously received words can be corrected, which reduces the number of retransmissions. However, in time-varying channels, these schemes lack flexibility in adapting to changing channel conditions: the additional parity bits for error correction may represent a waste of bandwidth when the channel BER is low for long periods of time; on the other hand, the designed error-correcting capability may not be sufficient for the occasional noisy periods.

2.5.2 HARQ-II Schemes

The adaptivity which is desired in time-varying channel environments is achieved to some extent by HARQ-II schemes, where the parity bits for error correction are sent only when they are needed. This is known as the method of incremental redundancy, the concept of which was first introduced by Mandelbaum in [51]. On the first transmission attempt, only parity bits for error detection are appended to the message, in the same way as in basic ARQ schemes. If errors are detected in the received word, it is stored in a buffer and a retransmission is requested. The retransmission is not the original codeword but a block of parity-check bits formed based on the original message and an error-correcting code. When this block is received, it is used to correct the errors in the previously stored erroneous word. If the error correction fails, another retransmission is requested, which can be either a repetition of the original codeword or another parity block, depending on the retransmission strategy and the type of error-correcting code that is used. This process continues until the original codeword is delivered successfully. One of the first articles to describe a scheme using incremental redundancy was [53], where Reed-Muller or convolutional codes were used for error correction.

Probably the most widely known HARQ-II scheme, and the first scheme to be called by that name, was proposed and analyzed in a BSC environment in [45] and improved in [74]. In this scheme, a rate-1/2 invertible block code or a rate-1/2 convolutional code is used for error correction. In the retransmissions, the original codeword and the parity block alternate, but only two packets are combined at a time to retrieve the original message. Throughput analysis of this scheme in a packet-level G-E channel was done in [48]. In [78], Yang and Bhargava studied the delay and coding gain performance of a ‘truncated’ HARQ-II scheme where at most one retransmission per packet is allowed, and in [50], Malkamäki and Leib studied the performance of truncated HARQ-II schemes in block fading channels.

The generalization of this idea to combination of more than two packets is known as code combining and was first suggested by Chase in [17], and the throughput performance of an HARQ-

II scheme using code combining was studied by Kallel in [34]. This type of schemes are often called *generalized HARQ-II schemes*. In [57], Mukhtar et al. analyzed both the throughput and the delay performance of a scheme with three ‘stages’ of code combining, and more general expressions for the mean transfer delay of an M-stage generalized HARQ-II scheme were obtained in [41] by using signal flow graphs.

In the recent years, numerous HARQ-II schemes have been proposed using advanced coding techniques, such as trellis-coded modulation [23], turbo coding [58] and zigzag codes [16].

If the packet combining is done after the quantization of the received data into bits, we have a *hard combining* system. Lately, there has been a substantial interest in HARQ schemes using *soft combining* methods. In [31], for example, the authors study a scheme where several erroneously received copies of a codeword are concatenated (without hard symbol quantization into bits) to form a noise-corrupted codeword in a longer, lower-rate code. The proposed soft combining method is obtained by using now the symbol-by-symbol MAP (maximum a posteriori) decoder for the aforementioned longer code.

2.5.3 Diversity Combining

Besides code combining, which is used in HARQ-II schemes, an alternative way of combining packets is to use diversity combining, where multiple identical (except for the errors) copies of a packet are combined to locate the errors [77, p. 394]. One such technique is to compute the bitwise modulo-2 sum (or logical XOR) of two received erroneous copies of a packet and to use the resulting ‘joint bit error map’ to retrieve the original message. This method was first proposed in [67], and the throughput analysis of an ARQ scheme with packet combining based on this idea was presented in [P1]. A ‘softer’ form of Sindhu’s combining method was proposed by Benelli in [7], where four-level quantization was performed on the received packets before combining. In [1], Adachi et al. studied a time diversity ARQ scheme which used maximal ratio combining (MRC) at the receiver, and in [76], Wicker added a majority-logic diversity combiner to a HARQ-I scheme to reduce the retransmissions.

2.6 Adaptive ARQ

By an adaptive ARQ scheme, we mean an ARQ scheme with two or more different transmission modes meant for different channel conditions, which uses some channel sensing mechanism to decide which transmission mode is used. A change of transmission mode can mean, for example, a change of the packet size in the SR scheme (e.g., [52]), or a change of the number of transmitted copies of a packet in the GBN scheme (e.g., [79]), or a change of the code rate in an HARQ-I scheme (e.g., [60]). In these schemes, the channel sensing is usually done by observing the acknowledgements sent by the receiver to the transmitter. This can mean either estimation of error rates, as in [52], or detection of channel state changes, as in [61] and [79], which does not require as long an observation interval (OBI) as reliable error rate estimation.

In [52], an adaptive SR scheme was proposed, where the packet size used in the current transmission was selected from a finite set of values based on a long-term BER estimate. This estimate was obtained by counting the incorrectly received packets over a time interval and assuming that there can be at most one bit error in an erroneous packet. Another adaptive SR scheme with variable packet size was proposed in [55], where the *a posteriori* distribution of the BER was computed based on the number of retransmissions during the OBI, and the packet size was selected so that the expected efficiency of the protocol was maximized. In [29], an adaptive SW scheme with variable packet size was proposed and simulated in a fading channel environment. The selection of the packet size was based on the PER estimate obtained by observing the acknowledgements over an OBI.

In [79], Yao proposed an adaptive GBN scheme where the transmitted number of copies of a packet was variable. The channel sensing algorithm suggested by Yao is used also in [P2]–[P6] and will be described in Section 3.1. Another adaptive GBN scheme was proposed in [39]. In this scheme, there are N transmission modes corresponding to the numbers of transmitted copies $1, \ldots, N$. The transmission mode is changed when a possible change of the channel state is detected.

Numerous adaptive HARQ schemes have been suggested in the literature. Typically, the code rate is varied according to the estimated channel conditions. In [72] and [73], adaptive HARQ-I schemes were studied with convolutional codes used for error correction. Finite-state Markov models were assumed for the channel. Switching between transmission modes depended on the number of erroneous blocks occurring during an OBI. A similar adaptive HARQ-I scheme with either block or convolutional codes was proposed in [60]. In [61], sequential statistical tests were applied on the acknowledgements to detect channel state changes. An adaptive HARQ-II scheme with variable packet size was proposed for wireless ATM networks in [33]. This scheme used rate compatible convolutional (RCC) codes for error correction. In [19], three different adaptive HARQ schemes are proposed using Reed-Solomon codes for error correction. Another adaptive HARQ scheme using Reed-Solomon codes with variable rate for error correction was proposed in [54]. In this scheme, short-term symbol error rate was estimated by computing the bitwise modulo-2 sum of two erroneous copies of a packet. This method was originally proposed in [15].

Chapter 3

Adaptive ARQ Schemes Based on Yao's Algorithm

3.1 Yao's Channel Sensing Algorithm

In [79], Yao proposed an adaptive GBN scheme with two transmission modes, L and H, meant for 'good' and 'bad' channel conditions, respectively. Mode L is the standard GBN scheme, but in mode H, m copies of the packet are sent at each transmission attempt. Switching between transmission modes is done based on the following simple algorithm: in mode L, if the transmitter receives α contiguous NACKs, it switches to mode H and begins multicopy transmissions. If the transmitter receives β contiguous ACKs in mode H, it switches immediately back to mode L.

3.2 Related Work

It was noted in [3, 13] that the simple two-state Markov chain used by Yao [79] in his analysis did not model the dynamics of the adaptive GBN scheme with sliding OBIs correctly, even in a stationary channel. Instead, a Markov chain with $\alpha + \beta$ states was needed. In [12], a slightly different adaptive GBN scheme with static OBIs of lengths α and β was modelled by a two-state semi-Markov process, still assuming a stationary channel environment. In [4], Yao's algorithm was applied to an adaptive SR scheme with variable packet size in stationary fading channels. Optimization of α and β for stationary channels has been studied in [3, 5] for an adaptive GBN scheme in a BSC environment, and in [4] for an adaptive SR scheme in a fading channel environment using the threshold model.

3.3 System Model and Throughput Analysis

In publications [P2]–[P6] of this thesis, the dynamics of the system consisting of the time-varying channel environment and the adaptive ARQ scheme are modelled in a straightforward fashion by Markov chains. The states of these processes are characterized by the state of the forward channel (G or B), (possibly) the state of the return channel (g or b), the transmission mode (L or H), and the state of the counter of contiguous NACKs in mode L ($0, \ldots, \alpha - 1$) or the counter of contiguous ACKs in mode H ($0, \ldots, \beta - 1$). The number of the states depends on the feedback channel model and on the values of the design parameters α and β.

In [P2]–[P5], we have studied adaptive SR schemes, were packet sizes of n_1 and n_2 bits, where $n_1 = 2n_2$, are used in transmission modes L and H, respectively. The number of parity bits for error detection, h, is the same in both the cases. The receiver sends the acknowledgements to the transmitter always after receiving n_1 bits. In the H mode, this means acknowledging pairs of n_2-bit packets; still, only the incorrectly received n_2-bit packets are retransmitted. These definitions make the scheme easy to implement. Switching between transmission modes is done based on a slightly modified version of Yao's algorithm: in mode L, if the transmitter receives α NACKs contiguously, it switches immediately to mode H; in mode H, after β contiguous pairs of n_2-bit packets have been received completely free of errors and acknowledged positively, mode L is resumed. In the simplest case, as in [P2] and [P3], the return channel is assumed to be error-free and the round-trip delay is assumed to be negligible. Then the system model has $2(\alpha + \beta)$ states, which are defined as follows:

(i) States $1, \ldots, \alpha$, also denoted by GL_r, where $r = 0, \ldots, \alpha - 1$: the channel state is G, the transmission mode is L, and the transmitter has received r contiguous NACKs. These states form the group GL.

(ii) States $\alpha + 1, \ldots, \alpha + \beta$, also denoted by GH_r, where $r = 0, \ldots, \beta - 1$: the channel state is G, the transmission mode is H, and the transmitter has received r contiguous 'double ACKs'. These states form the group GH.

(iii) States $\alpha + \beta + 1, \ldots, 2\alpha + \beta$, also denoted by BL_r, where $r = 0, \ldots, \alpha - 1$: these states form the group BL and are similar to the states GL_r, except that the channel is in state B.

(iv) States $2\alpha + \beta + 1, \ldots, 2(\alpha + \beta)$, also denoted by BH_r, where $r = 0, \ldots, \beta - 1$: these states form the group BH and are similar to the states GH_r, except that the channel is in state B.

The non-zero transition probabilities are given in Table 3.1. In the table entries, P_{CG} and P_{CB} denote the probabilities of a correct transmission in states G and B, respectively. The transition probabilities of the forward channel from G to B and from B to G are denoted by γ and δ, respectively, as was mentioned earlier in Section 2.1.1.

Figure 3.1 shows the state transition diagram for this system model when $\alpha = 2$ and $\beta = 3$.

Table 3.1: The non-zero transition probabilities of the system model in [P2]

old state	new state	probability
$GL_i,\ 0 \le i \le \alpha - 2$	GL_0	$(1-\gamma)P_{CG}$
	GL_{i+1}	$(1-\gamma)(1-P_{CG})$
	BL_0	γP_{CG}
	BL_{i+1}	$\gamma(1-P_{CG})$
$GL_{\alpha-1}$	GL_0	$(1-\gamma)P_{CG}$
	GH_0	$(1-\gamma)(1-P_{CG})$
	BL_0	γP_{CG}
	BH_0	$\gamma(1-P_{CG})$
$GH_i,\ 0 \le i \le \beta - 2$	GH_0	$(1-\gamma)(1-P_{CG})$
	GH_{i+1}	$(1-\gamma)P_{CG}$
	BH_0	$\gamma(1-P_{CG})$
	BH_{i+1}	γP_{CG}
$GH_{\beta-1}$	GL_0	$(1-\gamma)P_{CG}$
	GH_0	$(1-\gamma)(1-P_{CG})$
	BL_0	γP_{CG}
	BH_0	$\gamma(1-P_{CG})$
$BL_i,\ 0 \le i \le \alpha - 2$	GL_0	δP_{CB}
	GL_{i+1}	$\delta(1-P_{CB})$
	BL_0	$(1-\delta)P_{CB}$
	BL_{i+1}	$(1-\delta)(1-P_{CB})$
$BL_{\alpha-1}$	GL_0	δP_{CB}
	GH_0	$\delta(1-P_{CB})$
	BL_0	$(1-\delta)P_{CB}$
	BH_0	$(1-\delta)(1-P_{CB})$
$BH_i,\ 0 \le i \le \beta - 2$	GH_0	$\delta(1-P_{CB})$
	GH_{i+1}	δP_{CB}
	BH_0	$(1-\delta)(1-P_{CB})$
	BH_{i+1}	$(1-\delta)P_{CB}$
$BH_{\beta-1}$	GL_0	δP_{CB}
	GH_0	$\delta(1-P_{CB})$
	BL_0	$(1-\delta)P_{CB}$
	BH_0	$(1-\delta)(1-P_{CB})$

The performance of an adaptive ARQ scheme is measured here by its average throughput, which is defined as the average of the throughput of the scheme over all the states of the system model. For example, the average throughput of the adaptive SR scheme in [P2] is given by

$$\begin{aligned} \eta_{adapt} =& \eta_{SR}(n_1, \epsilon_1, h) \cdot \sum_{i \in GL} \pi_i + \eta_{SR}(n_2, \epsilon_1, h) \cdot \sum_{i \in GH} \pi_i + \\ & \eta_{SR}(n_1, \epsilon_2, h) \cdot \sum_{i \in BL} \pi_i + \eta_{SR}(n_2, \epsilon_2, h) \cdot \sum_{i \in BH} \pi_i\,, \end{aligned} \tag{3.1}$$

where for example, $\sum_{i \in GL} \pi_i$ is the probability that the forward channel is in state G and the transmission mode is L, and $\eta_{SR}(n_1, \epsilon_1, h)$ is the corresponding throughput, and so on.

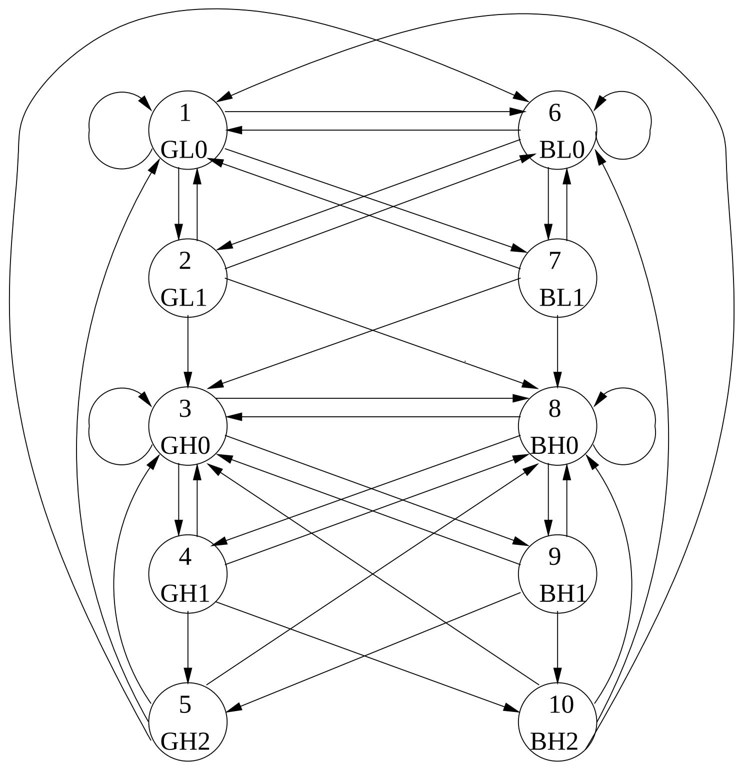

Figure 3.1: The state transition diagram of the system model in [P2] and [P3] when $\alpha = 2$ and $\beta = 3$

The average throughput of a 'real' scheme (specified by the values of α and β) is upper-bounded by that of an ideal adaptive scheme, which is always in the 'correct' transmission mode. That is, the time periods when the forward channel is in state G coincide with those when the transmission mode is L. In [P2], this upper bound is given by

$$\eta_{ideal} = \pi_G \cdot \eta_{SR}(n_1, \epsilon_1, h) + \pi_B \cdot \eta_{SR}(n_2, \epsilon_2, h)\,, \tag{3.2}$$

where the function $\eta_{\mathrm{SR}}(n, \epsilon, h)$ was defined in (2.10), and π_G and π_B are the steady-state probabilities of the forward channel being in states G and B, respectively.

In [P4], the round-trip delay was also assumed to be negligible, but a G-E model was now assumed also for the return channel. The state of the total system consisting of the adaptive SR scheme and the two channels develops in time according to a Markov chain with $4(\alpha + \beta)$ states, which are defined as follows:

(i) States $1, \ldots, \alpha$, also denoted by GgL_r, where $r = 0, \ldots, \alpha - 1$: the forward and return channel states are G and g, respectively, the transmission mode is L, and the transmitter has received r contiguous NACKs. These states form the group GgL.

(ii) States $\alpha + 1, \ldots, \alpha + \beta$, also denoted by GgH_r, where $r = 0, \ldots, \beta - 1$: the forward and return channel states are G and g, respectively, the transmission mode is H, and the transmitter has received r contiguous double-ACKs. These states form the group GgH.

(iii) States $\alpha+\beta+1,\ldots,2\alpha+\beta$, also denoted by GbL_r, where $r=0,\ldots,\alpha-1$: these states form the group GbL and are similar to the states GgL_r, except that the return channel is in state b.

(iv) States $2\alpha+\beta+1,\ldots,2(\alpha+\beta)$, also denoted by GbH_r, where $r=0,\ldots,\beta-1$: these states form the group GbH and are similar to the states GgH_r, except that the return channel is in state b.

(v) States $2(\alpha+\beta)+1,\ldots,3\alpha+2\beta$, also denoted by BgL_r, where $r=0,\ldots,\alpha-1$: these states form the group BgL and are similar to the states GgL_r, except that the forward channel is in state B.

(vi) States $3\alpha+2\beta+1,\ldots,3(\alpha+\beta)$, also denoted by BgH_r, where $r=0,\ldots,\beta-1$: these states form the group BgH and are similar to the states GgH_r, except that the forward channel is in state B.

(vii) States $3(\alpha+\beta)+1,\ldots,4\alpha+3\beta$, also denoted by BbL_r, where $r=0,\ldots,\alpha-1$: these states form the group BbL and are similar to the states GbL_r, except that the forward channel is in state B.

(viii) States $4\alpha+3\beta+1,\ldots,4(\alpha+\beta)$, also denoted by BbH_r, where $r=0,\ldots,\beta-1$: these states form the group BbH and are similar to the states GbH_r, except that the forward channel is in state B.

The transitions between the states of the system model happen at regular time intervals corresponding to n_1 transmitted bits. They are determined by

- state transitions of the two channels,
- success/failure of the transmission of the current packet,
- correctness of the corresponding received, acknowledgement (this is relevant only when the return channel is in state b).

The number of possible state transitions from one state of the system model is either 8 or 12, depending on the possibility and consequences of a feedback error. Note that in the preceding case of error-free return channel, there were only 4 possible transitions from each state.

It is very difficult to draw a complete state transition diagram of the system model clearly, even with small α and β. However, there is a lot of symmetry in the model. The set of states can be divided into four subsets in a very natural way based on the channel state combinations. We denote these subsets by Gg, Gb, Bg and Bb.

Table 3.2 shows the non-zero transition probabilities for two groups of states, GgL and BbH. In the table entries, $P_{\mathrm{e},1}$ and $P_{\mathrm{e},2}$ are the probabilities that there is at least one bit error in a

Table 3.2: The non-zero transition probabilities for two groups of states of the system model in [P4].

old state	new state	probability
$GgL_i,\ 0 \le i \le \alpha - 1$	GgL_0	$(1-\gamma)\,(1-\lambda)\,(1-P_{\mathrm{e},1})$
	GgL_{i+1} (if $i \ne \alpha - 1$)	$(1-\gamma)\,(1-\lambda)\,P_{\mathrm{e},1}$
	GgH_0 (if $i = \alpha - 1$)	$(1-\gamma)\,(1-\lambda)\,P_{\mathrm{e},1}$
	GbL_0	$(1-\gamma)\,\lambda\,(1-P_{\mathrm{e},1})$
	GbL_{i+1} (if $i \ne \alpha - 1$)	$(1-\gamma)\,\lambda\,P_{\mathrm{e},1}$
	GbH_0 (if $i = \alpha - 1$)	$(1-\gamma)\,\lambda\,P_{\mathrm{e},1}$
	BgL_0	$\gamma\,(1-\lambda)\,(1-P_{\mathrm{e},1})$
	BgL_{i+1} (if $i \ne \alpha - 1$)	$\gamma\,(1-\lambda)\,P_{\mathrm{e},1}$
	BgH_0 (if $i = \alpha - 1$)	$\gamma\,(1-\lambda)\,P_{\mathrm{e},1}$
	BbL_0	$\gamma\,\lambda\,(1-P_{\mathrm{e},1})$
	BbL_{i+1} (if $i \ne \alpha - 1$)	$\gamma\,\lambda\,P_{\mathrm{e},1}$
	BbH_0 (if $i = \alpha - 1$)	$\gamma\,\lambda\,P_{\mathrm{e},1}$
$BbH_i,\ 0 \le i \le \beta - 1$	GgL_0 (if $i = \beta - 1$)	$\delta\,\mu\,(1-P_{\mathrm{f}})\,(1-P_{\mathrm{e},2})$
	GgH_0 (if $i = 0$)	$\delta\,\mu\,[P_{\mathrm{f}} + (1-P_{\mathrm{f}})\,P_{\mathrm{e},2}]$
	GgH_0 (if $i \ne 0$)	$\delta\,\mu\,(1-P_{\mathrm{f}})\,P_{\mathrm{e},2}$
	GgH_i (if $i \ne 0$)	$\delta\,\mu\,P_{\mathrm{f}}$
	GgH_{i+1} (if $i \ne \beta - 1$)	$\delta\,\mu\,(1-P_{\mathrm{f}})\,(1-P_{\mathrm{e},2})$
	GbL_0 (if $i = \beta - 1$)	$\delta\,(1-\mu)\,(1-P_{\mathrm{f}})\,(1-P_{\mathrm{e},2})$
	GbH_0 (if $i = 0$)	$\delta\,(1-\mu)\,[P_{\mathrm{f}} + (1-P_{\mathrm{f}})\,P_{\mathrm{e},2}]$
	GbH_0 (if $i \ne 0$)	$\delta\,(1-\mu)\,(1-P_{\mathrm{f}})\,P_{\mathrm{e},2}$
	GbH_i (if $i \ne 0$)	$\delta\,(1-\mu)\,P_{\mathrm{f}}$
	GbH_{i+1} (if $i \ne \beta - 1$)	$\delta\,(1-\mu)\,(1-P_{\mathrm{f}})\,(1-P_{\mathrm{e},2})$
	BgL_0 (if $i = \beta - 1$)	$(1-\delta)\,\mu\,(1-P_{\mathrm{f}})\,(1-P_{\mathrm{e},2})$
	BgH_0 (if $i = 0$)	$(1-\delta)\,\mu\,[P_{\mathrm{f}} + (1-P_{\mathrm{f}})\,P_{\mathrm{e},2}]$
	BgH_0 (if $i \ne 0$)	$(1-\delta)\,\mu\,(1-P_{\mathrm{f}})\,P_{\mathrm{e},2}$
	BgH_i (if $i \ne 0$)	$(1-\delta)\,\mu\,P_{\mathrm{f}}$
	BgH_{i+1} (if $i \ne \beta - 1$)	$(1-\delta)\,\mu\,(1-P_{\mathrm{f}})\,(1-P_{\mathrm{e},2})$
	BbL_0 (if $i = \beta - 1$)	$(1-\delta)\,(1-\mu)\,(1-P_{\mathrm{f}})\,(1-P_{\mathrm{e},2})$
	BbH_0 (if $i = 0$)	$(1-\delta)\,(1-\mu)\,[P_{\mathrm{f}} + (1-P_{\mathrm{f}})\,P_{\mathrm{e},2}]$
	BbH_0 (if $i \ne 0$)	$(1-\delta)\,(1-\mu)\,(1-P_{\mathrm{f}})\,P_{\mathrm{e},2}$
	BbH_i (if $i \ne 0$)	$(1-\delta)\,(1-\mu)\,P_{\mathrm{f}}$
	BbH_{i+1} (if $i \ne \beta - 1$)	$(1-\delta)\,(1-\mu)\,(1-P_{\mathrm{f}})\,(1-P_{\mathrm{e},2})$

received n_1-bit sequence when the forward channel is in state G or B, respectively. They are given by

(3.3) $$P_{\mathrm{e},1} = P_{\mathrm{e}}(n_1, \epsilon_1)\,,$$

(3.4) $$P_{\mathrm{e},2} = P_{\mathrm{e}}(n_1, \epsilon_2)\,,$$

where $P_{\mathrm{e}}(n, \epsilon)$ was defined in (2.1).

Figure 3.2 shows the possible state transitions in a restricted model where only subsets Gg and Gb are included, when $\alpha = 2$ and $\beta = 3$. The model is symmetric in the sense that, for example from the states in subset Gb, exactly similar state transitions are possible to subsets Bg and Bb as to subset Gg.

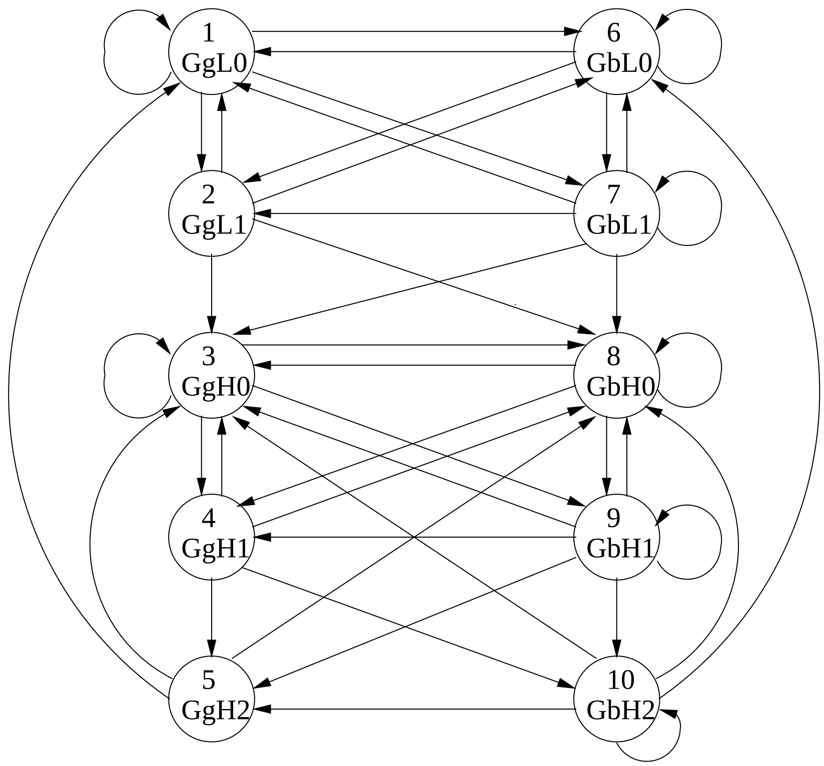

Figure 3.2: A partial state transition diagram of the system model in [P4] when $\alpha = 2$ and $\beta = 3$

In [P5] and [P6], it is no longer assumed that the round-trip delay is negligible. This makes it more difficult to model the system dynamics because the acknowledgement that arrives at the transmitter is determined by the outcome of the packet transmission that took place one round-trip delay ago; hence this acknowledgement depends statistically on the channel state *one round-trip delay ago.* A straightforward Markov model of this kind of system was presented in [14], and the number of states in the system was seen to grow exponentially with the delay parameter N.

A different, indirect method, which avoids the ‘state explosion’, was used in [P5]–[P6]. In [P6], for example, where an adaptive GBN scheme similar to the one in Yao’s original paper [79] was studied in an environment where a G-E model was assumed for both forward and return channels, a Markov chain with the following $4(\alpha + \beta)$ states was considered. Only the first two groups of states are described in detail; the others are defined in a similar manner.

(i) States $1, \ldots, \alpha$, also denoted by $G'gL_i$, where $i = 0, \ldots, \alpha - 1$: the forward channel was in state G $N-1$ time units ago, the return channel is in state g, the transmitter is in mode L and has received i contiguous NACKs. These states form the group $G'gL$.

(ii) States $\alpha+1,\dots,\alpha+\beta$, also denoted by $G'gH_i$, where $i=0,\dots,\beta-1$: same as $G'gL_i$, except that the transmitter is in mode H and has received i contiguous ACKs. These states form the group $G'gH$.

(iii) States $\alpha+\beta+1,\dots,2\alpha+\beta$, also denoted by $G'bL_i$, where $i=0,\dots,\alpha-1$, form the group $G'bL$.

(iv) States $2\alpha+\beta+1,\dots,2(\alpha+\beta)$, also denoted by $G'bH_i$, where $i=0,\dots,\beta-1$, form the group $G'bH$.

(v) States $2(\alpha+\beta)+1,\dots,3\alpha+2\beta$, also denoted by $B'gL_i$, where $i=0,\dots,\alpha-1$, form the group $B'gL$.

(vi) States $3\alpha+2\beta+1,\dots,3(\alpha+\beta)$, also denoted by $B'gH_i$, where $i=0,\dots,\beta-1$, form the group $B'gH$.

(vii) States $3(\alpha+\beta)+1,\dots,4\alpha+3\beta$, also denoted by $B'bL_i$, where $i=0,\dots,\alpha-1$, form the group $B'bL$.

(viii) States $4\alpha+3\beta+1,\dots,4(\alpha+\beta)$, also denoted by $B'bH_i$, where $i=0,\dots,\beta-1$, form the group $B'bH$.

The state transitions of this model are identical to those of the model in [P4]. The performance of the adaptive GBN scheme in [P6] was measured by the average packet throughput:

$$\begin{aligned} T_{ave} =& P(Gg'L)\,T_{Gg'L} + P(Gg'H)\,T_{Gg'H} + P(Gb'L)\,T_{Gb'L} + P(Gb'H)\,T_{Gb'H} + \\ & P(Bg'L)\,T_{Bg'L} + P(Bg'H)\,T_{Bg'H} + P(Bb'L)\,T_{Bb'L} + P(Bb'H)\,T_{Bb'H}\,, \end{aligned} \tag{3.5}$$

where for example $P(Gg'L)$ is the probability that the forward channel state is G and the transmitter is in mode L during the transmission of a packet, and the feedback channel is in state g when the corresponding acknowledgement arrives, i.e., $N-1$ time units later, while $T_{Gg'L}$ is the corresponding packet throughput.

The probability $P(Gg'L)$ can be expressed as

$$P(Gg'L) = \sum_{i=0}^{\alpha-1} P(Gg'L_i), \tag{3.6}$$

where the subscript i refers to the state of the counter of contiguous NACKs in mode L.

Note that the probabilities $P(Gg'L_i)$ are not given directly by the stationary distribution of the Markov chain presented above. Instead, they are obtained from the following simple calculation, which utilises the Markovian character and independence of the two channel models, and some basic rules of probability calculus (the total probability of an event).

$$\begin{aligned} P(Gg'L_i) =& P(G'gL_i)(1-\gamma^{(N-1)})(1-\lambda^{(N-1)}) + P(G'bL_i)(1-\gamma^{(N-1)})\,\mu^{(N-1)} + \\ & P(B'gL_i)\,\delta^{(N-1)}(1-\lambda^{(N-1)}) + P(B'bL_i)\,\delta^{(N-1)}\mu^{(N-1)}, \\ & i=0,\dots,\alpha-1\,. \end{aligned} \tag{3.7}$$

The probabilities $P(Gg'H)$, $P(Gb'L)$, $P(Gb'H)$, $P(Bg'L)$, $P(Bg'H)$, $P(Bb'L)$ and $P(Bb'H)$ are obtained in a similar way. The corresponding throughput values are given by the following equations

$$\begin{aligned} T_{Gg'L} &= T_{\text{GBN}}(N, m_1, P_{e,1}, 0)\,, \\ T_{Gg'H} &= T_{\text{GBN}}(N, m_2, P_{e,1}, 0)\,, \\ T_{Gb'L} &= T_{\text{GBN}}(N, m_1, P_{e,1}, P_f)\,, \\ T_{Gb'H} &= T_{\text{GBN}}(N, m_2, P_{e,1}, P_f)\,, \\ T_{Bg'L} &= T_{\text{GBN}}(N, m_1, P_{e,2}, 0)\,, \\ T_{Bg'H} &= T_{\text{GBN}}(N, m_2, P_{e,2}, 0)\,, \\ T_{Bb'L} &= T_{\text{GBN}}(N, m_1, P_{e,2}, P_f)\,, \\ T_{Bb'H} &= T_{\text{GBN}}(N, m_2, P_{e,2}, P_f)\,. \end{aligned} \tag{3.8}$$

An upper bound for T_{ave} is provided by the average packet throughput of an ideal adaptive GBN scheme, where the time periods when the forward channel is in state G (B) coincide with those when the transmission mode is L (H). Since the forward and feedback channels are independent, this upper bound is given by

$$\begin{aligned} T_{\text{ideal}} =& \pi_G \pi_g T_{\text{GBN}}(N, m_1, P_{e,1}, 0) + \pi_G \pi_b T_{\text{GBN}}(N, m_1, P_{e,1}, P_f) + \\ & \pi_B \pi_g T_{\text{GBN}}(N, m_2, P_{e,2}, 0) + \pi_B \pi_b T_{\text{GBN}}(N, m_2, P_{e,2}, P_f)\,. \end{aligned} \tag{3.9}$$

3.4 Parameter Optimization

In [3, 4, 5], the authors presented optimization results for parameters α and β. An adaptive SR scheme with variable packet size was considered in [4] in a Rayleigh fading environment using the threshold model, whereas the two other papers discussed an adaptive GBN scheme in a channel environment where packet errors occur randomly. However, none of these articles used a real time-varying channel model.

A different approach was used in publications [P4]–[P6] of this thesis. In each paper, a G-E channel model was used to represent a time-varying channel environment. Further, optimization of the packet size was studied for adaptive SR schemes in [P4] and [P5].

3.4.1 Optimization of the Packet Size

Parameter optimization for the adaptive SR scheme in [P4] and [P5] was done in two steps. First, the packet sizes are optimized for the 'ideal' adaptive scheme, and then the optimal packet sizes are adopted when parameters α and β are being optimized.

In the 'two-dimensional' packet size optimization, we let the BER values in forward channel states G and B, ϵ_1 and ϵ_2, vary over intervals $[\epsilon_{1,a}, \epsilon_{1,b}]$ and $[\epsilon_{2,a}, \epsilon_{2,b}]$, respectively. The scheme uses two different packet sizes, n_1 and n_2, where $n_1 = 2n_2$. The smaller packet size n_2 is taken as the independent parameter, and the optimal value of n_2 is defined to be the one that maximizes the double integral

$$\int_{\epsilon_{1,a}}^{\epsilon_{1,b}} \int_{\epsilon_{2,a}}^{\epsilon_{2,b}} \eta_{\text{ideal}}(\epsilon_1, \epsilon_2; n_2)\, d\epsilon_1 d\epsilon_2,$$

while satisfying the condition $\epsilon_{1,b} < \epsilon_{co} < \epsilon_{2,a}$, where ϵ_{co}, the cross-over BER between packet sizes n_2 and $2n_2$, is obtained from Eq. 2.12. Maximization of the integral above is equivalent to maximizing

$$I_1(n_2) = \int_{\epsilon_{1,a}}^{\epsilon_{1,b}} \int_{\epsilon_{2,a}}^{\epsilon_{2,b}} \left[\pi_G \left(1 - \frac{h}{2n_2}\right)(1-\epsilon_1)^{2n_2} + \pi_B \left(1 - \frac{h}{n_2}\right)(1-\epsilon_2)^{n_2}\right] d\epsilon_1 d\epsilon_2. \tag{3.10}$$

In the 'one-dimensional' optimization, ϵ_1 is fixed and optimization is performed over a range of ϵ_2-values, and the integral to be maximized is one-dimensional:

$$I_2(n_2) = \int_{\epsilon_{2,\mathrm{a}}}^{\epsilon_{2,\mathrm{b}}} \left[\pi_G \left(1 - \frac{h}{2n_2}\right)(1-\epsilon_1)^{2n_2} + \pi_B \left(1 - \frac{h}{n_2}\right)(1-\epsilon_2)^{n_2}\right] d\epsilon_2\,. \tag{3.11}$$

In this case, we require that $\epsilon_1 < \epsilon_{\mathrm{co}} < \epsilon_{2,\mathrm{a}}$.

3.4.2 Optimization of α and β

In [P4], the following (one-dimensional) approach is used in the optimization of α and β. We use the value of n_2 that maximizes $I_2(n_2)$, and define the optimal (α, β)-combination as the one that minimizes the mean-square difference of η_{ave} and η_{ideal} over the interval $[\epsilon_{2,a}, \epsilon_{2,b}]$. This is approximately equivalent to minimizing the sum

$$E(\alpha, \beta) = \sum_{k=1}^{K} \left[\eta_{ave}(\epsilon_{2,k}; \alpha, \beta) - \eta_{\text{ideal}}(\epsilon_{2,k})\right]^2, \tag{3.12}$$

where the sample values $\epsilon_{2,k},\ k = 1, \ldots, K$, are evenly spaced on the interval $[\epsilon_{2,a}, \epsilon_{2,b}]$. Since we do not have a closed-form expression for η_{ave} as a function of α, β and other parameters, a computer search must be used in the optimization. Figure 3.3 shows the average throughput of the adaptive SR scheme in [P4] with different values of α and β, and the upper bound η_{ideal}, as functions of ϵ_2 (the BER in state B) for a given set of values of the parameters specifying the SR scheme. The difference between the ideal curve and the optimal actual performance curve corresponding to $(\alpha, \beta) = (2, 16)$ is at most points too small to be visible. The two other parameter combinations illustrated here are clearly inferior choices.

3.5 Summary of Optimization Results

The optimal value of n_2 is determined essentially by the proportion of time that the forward channel stays in state B. The smaller the value of π_B is, the bigger is the optimal value of n_2.

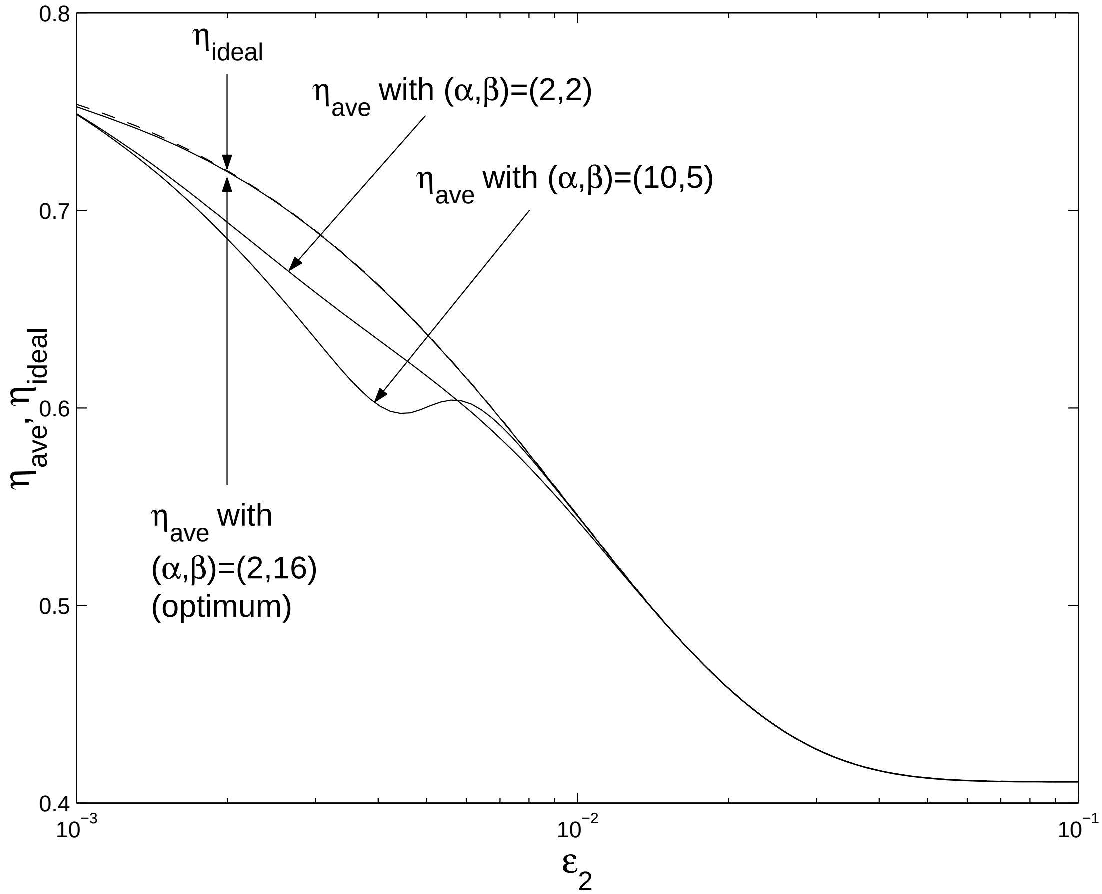

Figure 3.3: The performances of the ideal scheme and the adaptive SR scheme with different (α, β)-combinations when $\gamma_0 = \delta_0 = 10^{-6}$

It is hardly surprising that the optimal values of α and β are larger for more slowly varying forward channels. An increased value of the round-trip delay or the probability of feedback erasure tend to decrease the optimal values of β slightly, as can be seen in Table 3.3, where D_0 is the round-trip delay expressed as the number of bits that can be transmitted during the time.

Table 3.3: Optimization results when $P_f = 0$.

γ_0	δ_0	D_0	$(\alpha, \beta)_{\text{opt}}$
10^{-5}	10^{-5}	1000	$(2, 8)$
10^{-6}	10^{-6}	1000	$(3, 26)$
10^{-7}	10^{-7}	1000	$(3, 37)$
10^{-8}	10^{-8}	1000	$(4, 56)$
10^{-9}	10^{-9}	1000	$(4, 65)$
10^{-5}	10^{-5}	10000	$(2, 7)$
10^{-6}	10^{-6}	10000	$(2, 13)$
10^{-7}	10^{-7}	10000	$(3, 35)$
10^{-8}	10^{-8}	10000	$(4, 55)$
10^{-9}	10^{-9}	10000	$(4, 64)$

Chapter 4

ARQ with Diversity Combining

4.1 The EARQ Scheme with Time Diversity

In [67], Sindhu proposed a simple idea of combining incorrectly received packets to recover the correct packet. A crucial procedure in this scheme was the computation of the bit-wise XOR (Exclusive-OR) or, equivalently, the bit-wise modulo-2 sum of two incorrectly received versions of a packet. The purpose of this operation was to get information about the potential locations of the errors in the packets. The analysis in [67] considered only a burst-error channel environment. In [P1], this idea has been applied to a simple SR ARQ scheme resulting in what is called the *extended ARQ (EARQ)* scheme.

The EARQ scheme uses only one (n, k) code, which is used for error detection only. The scheme operates as follows. If errors are detected in the first transmission of a codeword, the received vector is stored in the receiver buffer and a retransmission is requested. If the retransmission is error-free, the received vector is assumed to be the original codeword and an ACK is sent to the transmitter. If errors are detected also in the retransmission, the bit-wise XOR of the two erroneous copies of the codeword is computed.

The output of the XOR operation is an n-bit vector with 0s in the bit positions where the copies coincide and 1s in the bit positions where they differ. The positions with 1s are the ones where exactly one of the copies has an error. If, on the other hand, there is at least one bit position where both copies have an error, this results in a 0 in the output of the XOR operation. This event is called a *double error* in [P1].

The next step is to try to recover the original codeword by a straightforward search procedure. This procedure begins from one of the two copies and starts to go through the potential error patterns: the corresponding bits are inverted and the syndrome of the resulting vector is computed based on the error-detecting code. This process continues until either a vector with zero syndrome has been obtained or all the possibilities have been checked without a positive result. In the first case, it is assumed that the correct codeword has been recovered and an ACK is sent to the transmitter, which proceeds to transmit the next codeword. In the second case, a double error has occurred and a NACK is sent requesting second retransmission of the codeword.

If the second retransmission is found to be error-free, it is accepted and an ACK is sent to the transmitter. If errors are detected, the packet combining procedure is carried out with, if necessary, both the earlier received erroneous copies. If one of these operations is successful, an ACK is sent. Otherwise, yet another retransmission is requested. This time, the new copy, if found erroneous, can be combined with three earlier copies, and so on. The process, of course, continues until the codeword has been received correctly or recovered by the packet combining procedure.

One aspect that deserves special attention is the *computational complexity* of the search process mentioned above. If there are n_1 1s in the output of the XOR operation, there are $n_p = 2^{n_1} - 2$ potential error patterns, all of which are checked if the search is unsuccessful. It is easy to see that n_p increases very rapidly with the noisiness of the channel. One solution to this problem is to define some limit $N_{\max}$, as in [P1]: packet combining is not attempted with a pair of packets if the output of the XOR operation contains more than $N_{\max}$ 1s.

4.2 Approximate Throughput Analysis

In [P1], the EARQ scheme is analyzed assuming a BSC with BER equal to p_e. Further, it is assumed that the round-trip delay is negligible and that the applied code provides perfect error detection. The packet throughput of the scheme is obtained from

$$T_{EARQ} = \frac{1}{E[L]}\,, \tag{4.1}$$

where the random variable L is defined as the number of transmissions required until a codeword has been successfully received, and its expectation value $E[L]$ is naturally obtained from

$$E[L] = \sum_{L=1}^{\infty} L \cdot P(L)\,. \tag{4.2}$$

Clearly, the probability that one transmission is sufficient is equal to $P_c = 1 - P_e$. For $L > 1$,

$$P(L) = \left[1 - \sum_{r=1}^{L-1} P(r)\right] \left[P_c + P_e(1 - \alpha_d(L))\right]\,, \tag{4.3}$$

where $\alpha_d(L)$ is the conditional probability that, given that all the L copies are erroneous and that all pairs taken out of the first $L - 1$ copies have double errors, the Lth copy has a double error with all the $L-1$ earlier copies. The exact calculation of $\alpha_d(L)$ gets complicated when $L \geq 3$. For example, when $L = 3$, the events of the pairs $(1,2)$, $(1,3)$ and $(2,3)$ having double errors are not independent. However, it turns out that the assumption of 'independent double errors' is a good approximation of reality, as demonstrated by the good agreement between the simulations and theoretical calculations in [P1]. Using the independence assumption, we get

$$\alpha_d(L) = \frac{\sum_{k_1=1}^{n} \cdots \sum_{k_L=1}^{n} P(k_1,\ldots,k_L) \cdot \prod_{\substack{1 \leq i,j \leq L \\ i<j}} P_{i,j}}{P_e^L \cdot \prod_{l=2}^{L-1} \alpha_d(l)}\,, \tag{4.4}$$

where

$$P(k_1,\ldots,k_L) = P(k_1)P(k_2)\cdot\cdot P(k_L) \tag{4.5}$$

is the joint probability of the L copies having $k_1,\ldots,k_L$ errors, respectively. Since the BSC model is assumed, the numbers of errors in the copies are independent and identically distributed random variables with binomial distributions:

$$P(k_i) = \binom{n}{k_i} p_e^{k_i}(1-p_e)^{n-k_i}\,. \tag{4.6}$$

Further, $P_{i,j}$ is the probability that two copies with k_i and k_j errors, respectively, have a double error. With the assumption that $k_1 + k_j \leq n$, it is easy to show that

$$P_{i,j} = 1 - \frac{(n-k_i)!(n-k_j)!}{n!(n-k_i-k_j)!}\,. \tag{4.7}$$

If the parameter $N_{\max}$ is also taken into account, the situation where there are more than $N_{\max}$ 1s in the output of the XOR operation has the same effect as a double error. Thus, the following modification is needed:

$$P_{i,j} = \begin{cases} 1 - \frac{(n-k_i)!(n-k_j)!}{n!(n-k_i-k_j)!}\,, & \text{if } k_i + k_j \leq N_{\max}, \\ 1, & \text{if } k_i + k_j > N_{\max}\,. \end{cases} \tag{4.8}$$

The 'limited' EARQ scheme has been simulated with three different values of $N_{\max}$ using 100-bit packets. Figure 4.1 provides the results from these simulations along with approximated theoretical packet throughputs for both 'limited' and 'unlimited' EARQ schemes. In this case, the probabilities $P(L)$ have been computed up to $L = 5$, and $P(6)$ has been approximated by the tail distribution. Even with $N_{\max} = 10$, packet throughput of approximately 33% can be achieved at BER as high as 0.05.

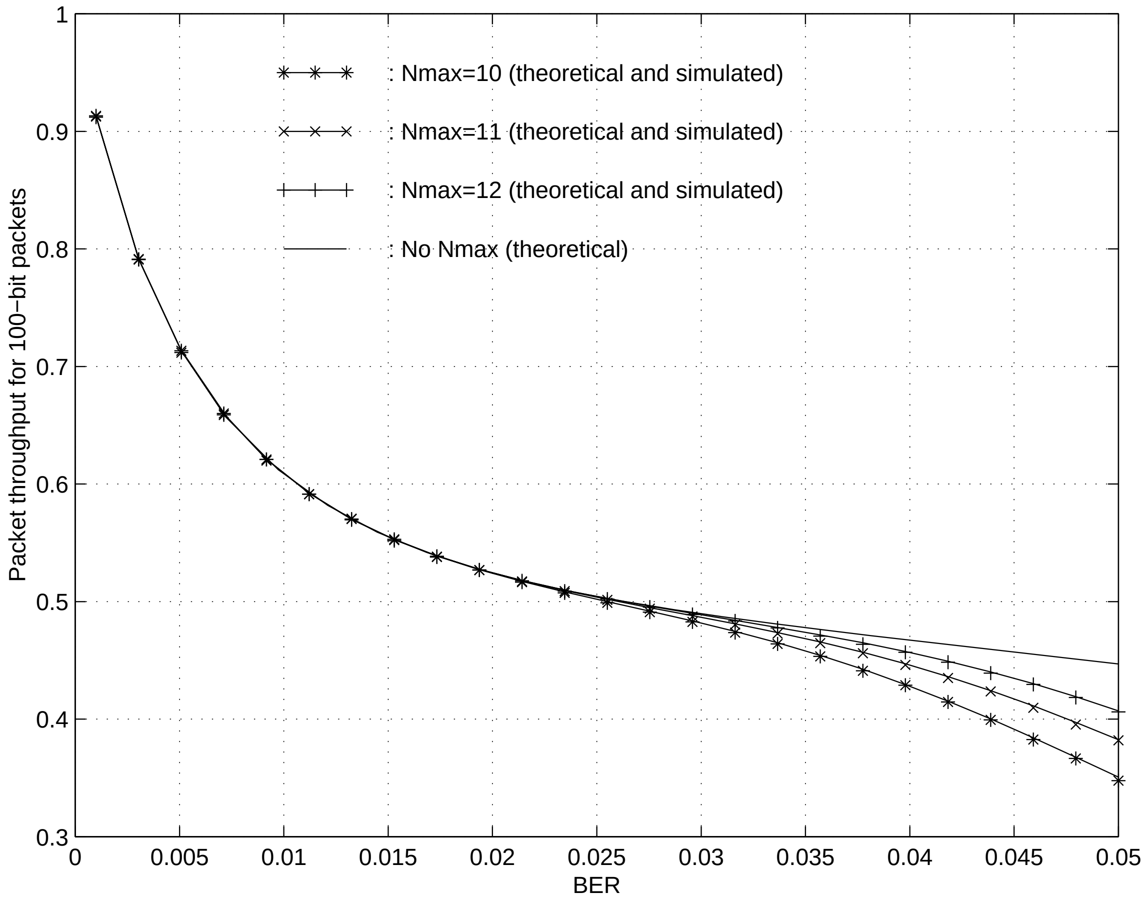

Figure 4.1: The packet thoughput of the EARQ scheme with and without $N_{\max}$

4.3 The EARQ Scheme with Spatial Diversity

The throughput performances of four different ARQ schemes are compared in the context of space diversity reception in [P7]: (i) the basic SR ARQ scheme (no diversity); (ii) the SR ARQ scheme with switched diversity, where only one receiving antenna is active at a time, and the active antenna is changed after every erroneously received packet (the ‘SAD’ scheme in the paper); (iii) the SR ARQ scheme with modified selection diversity, where both the antennas are always at use, and an ACK is sent to the transmitter if at least one of the antennas receives the packet without errors (the ‘NSD’ scheme in the paper); (iv) the EARQ scheme studied in [P1], this time utilizing spatial diversity and combining two erroneous copies of a packet that have been received simultaneously on the two antennas.

In the AWGN channel, the throughputs can be calculated analytically. If the BPSK modulation scheme is used, and the channel code rate is R_c, the bit error probability in an AWGN channel with the SNR equal to E_b/N_0 is given by [6]

$$p = \frac{1}{2}\,\mathrm{erfc}\left(\sqrt{\frac{E_b R_c}{N_0}}\right). \tag{4.9}$$

Note that, since bit errors are random in AWGN channel, packet errors are also random. Thus, the SAD scheme does not provide any benefit over simple SR scheme in AWGN channels. In the NSD scheme, the probability of successfully receiving a packet is that of a successful reception over either of the diversity branches, that is,

$$P_1 = 1 - (1 - (1 - p)^n)^2, \tag{4.10}$$

and the throughput is

$$\eta_{\mathrm{NSD}} = \frac{k}{n} P_1. \tag{4.11}$$

In the SR-NSD-EARQ scheme, when both copies are erroneous, a packet is retrieved correctly if the combined copy does not have a double error, and if the total number of errors in the combined copy is at most $N_{\max}$. The probability that both these conditions are satisfied is

$$\begin{aligned} P_2 &= \sum_{k_1=1}^{N_{\max}-1} \left\{ \binom{n}{k_1} p^{k_1} (1-p)^{n-k_1} \cdot (1-p)^{k_1} \sum_{k_2=1}^{N_{\max}-k_1} \left[\binom{n-k_1}{k_2} p^{k_2} (1-p)^{n-k_1-k_2} \right] \right\} \\ &= \sum_{k_1=1}^{N_{\max}-1} \sum_{k_2=1}^{N_{\max}-k_1} \binom{n}{k_1} \binom{n-k_1}{k_2} p^{k_1+k_2} (1-p)^{2n-k_1-k_2}, \end{aligned} \tag{4.12}$$

and the throughput of the EARQ scheme is given by

$$\eta_{\mathrm{EARQ}} = \frac{k}{n}(P_1 + P_2). \tag{4.13}$$

In Figure 4.2, the throughput curves of these schemes are plotted with $k = 84$ and $n = 100$. It can be seen that, even with $N_{max} = 4$, the EARQ scheme provides considerable performance gain over the other schemes with very reasonable additional computational complexity.

The performances of the four schemes in Rayleigh fading channels are compared by simulations using the Jakes model. The throughput curves of the four ARQ schemes as functions of the average SNR, when $v = 1\ m/s$, $n = 100$ bits, and $N_{\max} = 10$, are shown in Figure 4.3. One observation that can be made is that the SAD scheme is clearly superior to the basic SR scheme over the wide range of SNR considered here. This indicates that the average duration of fades is fairly long, and the switched antenna diversity improves the performance significantly. The SR-NSD and EARQ schemes can provide further performance improvements.

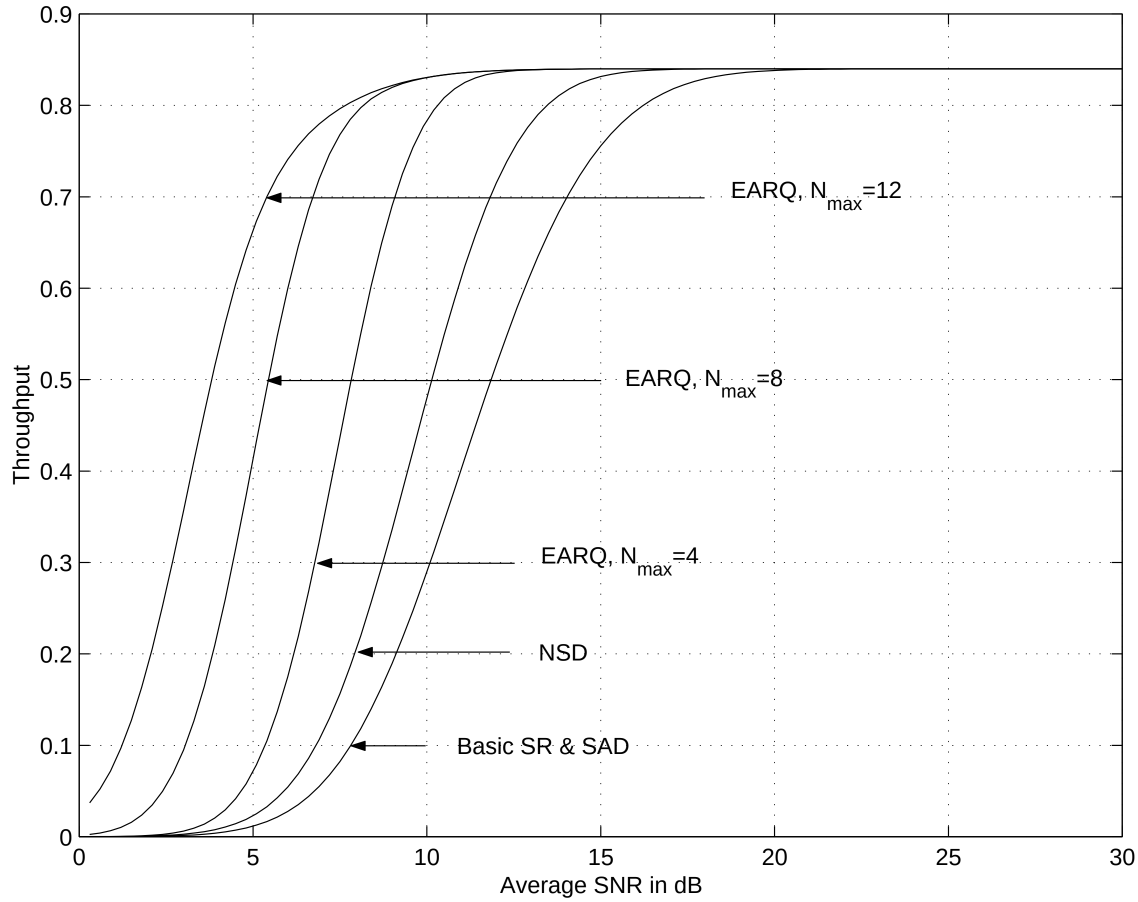

Figure 4.2: Throughput curves of four ARQ schemes in AWGN channel, with $k = 84$ and $n = 100$

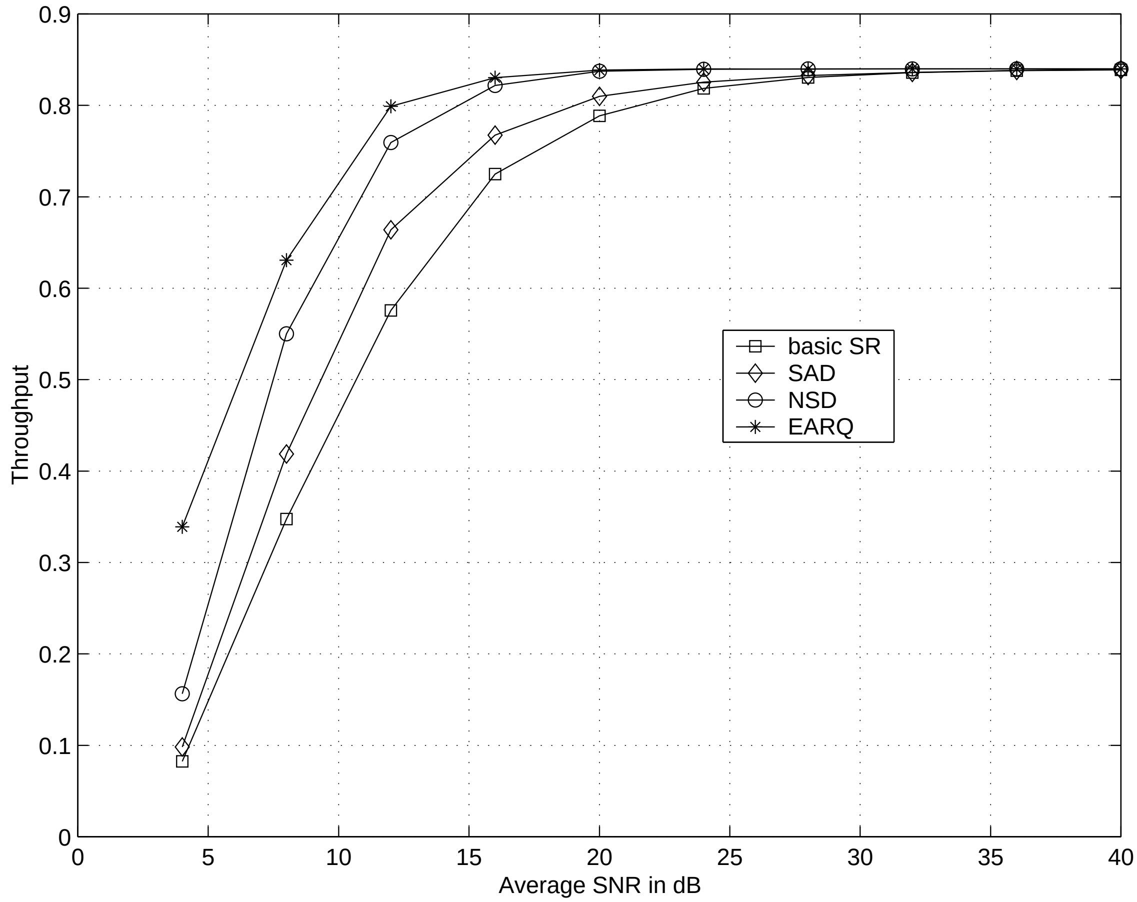

Figure 4.3: Simulated throughputs of the four ARQ schemes when the packet size is 100 bits, $v = 1\ m/s$, and $N_{\max} = 10$

Chapter 5

Summary of Publications

In [P1], we present an approximate (packet) throughput analysis and a simulation study of an ARQ scheme with packet combining in the stationary BSC environment. The packet combining algorithm, where the bitwise modulo-2 sum of two erroneous copies of a packet is computed to locate the errors, was first proposed by Sindhu in [67]. The validity of the approximate analysis is verified by the good agreement with the simulation results. The packet combining algorithm involves a straightforward search procedure, the computational complexity of which easily becomes prohibitive. As a solution to this, a modified scheme is proposed, where the search procedure is attempted only when there are at most $N_{\max}$ 1s at the output of the modulo-2 adder. Packet throughput curves of the modified scheme (both theoretical and simulated) as a function of the channel BER are shown with various values of $N_{\max}$.

Publication [P2] introduces an adaptive SR ARQ scheme with two transmission modes, mode L for 'good' and mode H for 'bad' channel conditions. In mode L, the packet size is n_1 and in mode H n_2 bits, where $n_1 = 2n_2$. The smaller packets are acknowledged in pairs by sending 'bitmaps' to the transmitter. The switching between the two transmission modes is done according to the algorithm introduced by Yao in [79]: in mode L, if the transmitter receives α NACKs contiguously, it switches immediately to mode H and starts to transmit smaller packets; if the transmitter is in mode H and receives β 'double ACKs' contiguously, it switches to mode L. The round-trip delay is assumed to be negligible, and the return channel is assumed to be error-free. The time-varying forward channel is represented by a packet-level G-E model, where the states G and B are defined by the BER values ϵ_1 and ϵ_2, respectively, where $\epsilon_2 \gg \epsilon_1$. The system is modelled by a Markov chain with $2(\alpha+\beta)$ states, and the average throughput of the adaptive SR scheme is obtained as the time average of the throughput over the states of the Markov model. As a basis of comparison, we define a hypothetical ideal adaptive scheme which knows the channel state and chooses the transmission mode accordingly: when the forward channel is in state G (B), the transmission mode is L (H). For all the values of the design parameters α and β, the average throughput of the adaptive scheme is upper-bounded by that of the ideal scheme.

In [P3], the same adaptive SR scheme is studied as in [P2], and the assumptions about the return channel and the round-trip delay are the same. However, the states of the forward channel model represent now Rayleigh fading channels defined by the values of the parameters ρ and

f_D according to the threshold model described in Section 2.1.2. The performance analysis is very similar to that presented in [P2].

The same adaptive SR scheme is also studied in [P4], and the round-trip delay is still assumed to be negligible. The time-varying forward channel is now represented by a bit-level G-E model so that the channel model is defined independently of the packet size, and this time a G-E model is assumed also for the return channel: state g is error-free, and in state b an acknowledgment is erased with probability P_f. The system model has now $4(\alpha + \beta)$ states. In this paper, we also study optimization of the 3 independent design parameters of the adaptive SR scheme: n_2, α and β. This is done in two steps. First we try to find the best possible value of n_2 by optimizing the performance of the ideal scheme. Then we fix n_2 and optimize α and β by minimizing (approximately) the mean-square difference between the ideal and actual performance curves (or 'surfaces' in the case of two-dimensional optimization). Probably the most important observation of this article is the importance of the judicious selection of the value of n_2 for the performance of the adaptive scheme.

Publication [P5] also studies analysis and optimization of the same adaptive SR scheme. The forward channel model is the same as in [P4], but now the acknowledgments are assumed to be erased randomly with probability P_f in the return channel. The important difference from the earlier articles is that the round-trip delay is no longer assumed to be negligible. The state-space explosion with the increasing delay, which was encountered by the authors in an earlier work [14], is avoided by using a modified Markov model, which has always $2(\alpha + \beta)$ states irrespective of the round-trip delay. The optimal value of β is seen to depend slightly on the values of the round-trip delay and P_f.

In [P6], an adaptive GBN scheme using Yao's algorithm is analyzed and optimized. The two transmission modes L and H correspond to m_1-copy and m_2-copy GBN schemes, respectively. In all the numerical examples, we have set $m_1 = 1$. A packet-level G-E model is assumed for the forward channel, where the states G and B are characterized by the PER values $P_{e,1}$ and $P_{e,2}$, with $P_{e,2} \gg P_{e,1}$. A similar model is assumed also for the return channel: state g is error-free, and in state b the acknowledgments are erased with probability P_f. The performance of the adaptive scheme is measured by its average packet throughput over the $4(\alpha + \beta)$ states of the system model, and the optimization of α and β is carried out in the same manner as in [P4] and [P5]. The paper also contains a brief discussion of how the burstiness of the forward channel errors affects the performance of the adaptive GBN scheme.

Publication [P7] is about ARQ and spatial diversity. The communication system consists of one transmitting antenna and two receiving antennas. The two diversity branches are assumed to behave like two independent flat Rayleigh fading channels, which are simulated by using the Jakes model. The BPSK modulation scheme is used with coherent demodulation. We compare the simulated throughput performance of four different ARQ schemes: (i) the basic SR ARQ scheme (no diversity); (ii) the SR ARQ scheme with switched diversity, where only one receiving antenna is active at a time, and the active antenna is changed after every erroneously received packet; (iii) the SR ARQ scheme with modified selection diversity, where both the antennas are always at use, and an ACK is sent to the transmitter if at least one of the antennas receives the packet without errors; (iv) the EARQ scheme studied in [P1], this time utilizing spatial diversity and combining two erroneous copies of a packet that have been received simultaneously on the two antennas. The theoretical throughput performances of the schemes in the AWGN channel are also compared. The main contribution of this article is to present the EARQ scheme as a potential method of choice for systems using spatial diversity reception.

Chapter 6

Conclusions

In publications [P2–P6] of this thesis, adaptive ARQ schemes based on Yao's channel sensing algorithm have been analyzed and optimized in Gilbert-Elliott channels. There are two kinds of non-ideal behaviour that make the throughput curves of the adaptive ARQ schemes deviate from those of the ideal schemes. Firstly, there is delay in reacting to changes in the channel state. This delay becomes of course longer if the values of α and β are bigger. Secondly, 'false alarms' occur, making the scheme switch between transmission modes unnecessarily. Here, bigger values of α and β make the scheme more 'reliable'. Neither of these phenomena can be avoided completely. Instead, they must be 'balanced' by finding the optimal (α, β)-combination based on the time-varying characteristics of the channel. The closest comparison is to the work of Annamalai and Bhargava in [3, 4, 5], where these authors did not specify any time-varying channel model, but analyzed the performance of the adaptive scheme in stationary channels. Their approach to the optimization of α and β was thus to minimize the degrading effect of 'false alarms' on the performance of the scheme. This appears to be the reason why the optimal solutions 'lie in the infinite $\alpha - \beta$ space', as the authors put it. Finite sub-optimal solutions were found only by introducing an artificial upper bound $\alpha_{\max}$ for the values of α . The articles [P4–P6] of this thesis presents a different approach to the optimization of the adaptive ARQ schemes based on Yao's algorithm for time-varying channels. In our case, finite optimal solutions exist, because too large values of α and β would make the scheme react too slowly to channel state changes. Our results demonstrate that the optimal parameter values depend strongly on the time-varying characteristics of the channel. An important new contribution of publications [P4–P5] is the optimization of the packet size(s) used by the adaptive SR scheme.

The EARQ scheme is an ARQ scheme with packet combining. In [P1], time diversity was utilized, whereas space diversity reception was considered in [P7]. In both cases, the scheme gives a considerable gain in throughput performance over the basic SR ARQ scheme. However, this gain is not achieved without cost; if the expected number of bit errors per block increases, the complexity of the search procedure associated with the combining algorithm quickly becomes prohibitive. To avoid this problem, the value of the parameter $N_{\max}$ must be selected judiciously, so that significant performance gain is still achieved but excessive processing delay at the receiver is avoided. An interesting further study would be to compare the performance of the EARQ scheme to that of some soft combining schemes, which preserve more information about the received symbols.

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