JOINT AMC-ARQ TRANSMISSION IN WIRELESS SYSTEMS WITH FINITE-LENGTH TRANSMIT BUFFERS

by

Sanal Sasankan
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SUPERVISORY COMMITTEE

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Supervisory Committee

Dr. Hong-Chuan Yang, Supervisor, (Department of Electrical and Computer Engineering)

Dr. Fayez Gebali, Member, Member, Department Member, (Department of Electrical and Computer Engineering)

Dr. Kui Wu, Member, Outside Member, Department Member (Department of Computer Science)

Dr. Sudhakar Ganti, External Examiner (Department of Computer Science)

Supervisory Committee

Supervisor Dr. Hong-Chuan Yang, Department of Electrical and Computer Engineering.

Co-Supervisor or Department Member: Dr. Fayez Gebali, Department of Electrical and Computer Engineering.

Department Member: Dr. Kui Wu, Department of Computer Science.

Department Member: Dr. Sudhakar Ganti, Department of Computer Science.

ABSTRACT

Adaptive modulation and coding (AMC) and automatic repeat request (ARQ) are two of the most important techniques to overcome the limitations of the wireless channel, and to thereby improve quality of service (QoS) over wireless networks. Traditionally, the performance of these two techniques have been designed and analyzed separately. In this thesis, we develop a novel cross-layer design of joint AMC-ARQ transmission for wireless systems and analyze its performance while taking into account the finite-length buffer constraint at the transmitter. This work differs from previous works in that it considers the effect of the transmitter buffer size and the different ARQ strategies in the design and performance analysis. In particular, a new buffer-assisted AMC strategy is presented in which the data rate is selected based on channel fading conditions as well as the status of the transmit buffer. We carry out detailed performance analysis of the proposed systems with different traffic arrival patterns and buffer management strategies, based on the Markov chain theory. Mathematical analysis and simulations show that an all retransmit (AR) retransmission strategy achieves the best performance and power efficiency amongst all retransmission strategies. The results also indicate that a partial retransmit (PR) strategy would provide a fair tradeoff between improved performance and increased receiver complexity. Further, we show that the buffer-assisted AMC technique can improve the transmission reliability when compared to the traditional channel-based AMC strategy.

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List of Abbreviations

AMC Adaptive Modulation and Coding

ARQ Automatic Repeat Request

AR All Retransmit

PR Partial Retransmit

SR Selective Retransmit

GSM Global System for Mobile Communication

GPRS General Packet Radio Service

EDGE Enhanced Data Rates for Global Evolution

UMTS Universal Mobile Communication System

HSDPA High-Speed Downlink Packet Access

TDMA Time Division Multiple Access

CDMA Code Division Multiple Access

FDD Frequency Division Duplexing

ARFCN Absolute Radio Frequency Channel Number

TCH Traffic Channel

CCH Control Channel

MCS Modulation and Coding Schemes

3GPP Third Generation Partnership Project

QPSK Quadrature Phase Shift Keying

QAM Quadrature Amplitude Modulation

IR Incremental Redundancy

A^2IR Adaptive Incremental Redundancy

UE User Equipment

CQF Channel Quality Feedback

H-ARQ Hybrid ARQ

SNR Signal-to-Noise Ratio

QoS Quality of Service

Acknowledgement

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Finally, I would like to thank my parents and sister whose support and encouragement have been invaluable throughout my career.

Dedication

To my parents and sister.

Chapter 1

Introduction

Wireless communication has been constantly evolving since Guglielmo Marconi demonstrated the wireless telegraph at the English telegraph office in 1896. This pioneering work was followed by key innovations in digital communications, cellular concept and coding theory. The advances in micro-electronic circuits and digital signal processing over the past two decades has enabled the rapid development and widespread usage of fixed and mobile wireless communication systems. Transmission of high-speed Internet and real-time video traffic over wireless channels with a quality of service comparable to that of wired networks will form the next major milestone in the evolution of wireless communications. Transmission of real-time traffic over the wireless channel is inherently challenging due to the deletrious effects of multipath fading, Doppler, and time-dispersion effects that limit the throughput performance of wireless systems. Due to multipath fading, the signal received at a receiver will contain not only a direct line-of-sight component, but also a large number of reflected radio waves. This leads to constructive and destructive interference that causes the amplitude and phase of the received signal to vary rapidly. In addition, the motion of a receiver with respect to the transmitter can result in a frequency shift in the received signal due to the Doppler effect. These effects cause the received signal strength to fluctuate widely depending on the channel fading conditions.

The use of a fixed, high-data rate modulation scheme at the wireless transmitter would result in higher error rates when the channel conditions are unfavorable, thus reducing the overall throughput of the system. Various techniques have been developed to improve the throughput and efficiency over wireless links. Adaptive transmission is one such technique implemented at the physical layer to achieve higher spectral efficiency by adapting the symbol transmission rate, constellation size or coding rate to the prevailing channel conditions. This technique allows the wireless transmitter to transmit at higher data rates when the channel conditions are conducive and vice versa, resulting in better utilization of the allocated frequency spectrum.

1.1 Adaptive Modulation and Coding

The quality of a signal received by a mobile terminal depends on a number of factors - the distance between the desired and interfering base stations, path loss exponent, log-normal shadowing, short term Rayleigh fading and noise. Fixed modulation schemes require a fixed link margin to maintain acceptable performance when the channel conditions are poor [1]. Therefore, these systems have to be designed for worst-case channel conditions. For example, Rayleigh fading can result in signal attenuation of up to 30dB, and designing the system to accomodate this power loss can result in very inefficient utilization of the wireless channel. Adaptive transmission provides the flexibility to match the modulation-coding scheme to the average channel quality or SNR for each user. The basic premise of adaptive transmission is to estimate the channel at the receiver and feed this estimate back to the transmitter, so that transmissions can be adapted relative to the channel characteristics. Adapting transmission to the channel fading conditions can result in increased throughput, less transmit power and less probability of bit error. Link adaptation can be achieved by varying several parameters [1]:

Variable-rate transmission In variable-rate transmission, the data rate of transmission is varied relative to the channel gain. The data rate can be varied by either keeping the symbol rate constant and using multiple modulation schemes, or by fixing the modulation and changing the symbol rate. The IS-136, EDGE as well as IEEE 802.11a wireless systems use variable-rate transmission for link adaptation.

Variable-power transmission The transmitted power alone can be adapted to compensate for SNR variation due to slow fading. This is done so that the system maintains a constant received SNR. The power adaptation thus inverts the channel fading so that the channel appears as an AWGN channel to the modulator and demodulator.

Variable-coding transmission Adaptive coding is used to vary the coding rate of the system to the channel fading conditions. A stronger error correction code is used when the SNR is small, and a weaker code is used when the SNR is large. Adaptive coding can be implemented by multiplexing codes with different error correction capabilities. However, this requires that the received SNR be constant over the fixed frame period. Rate-compatible punctured convolutional codes (RCPC) are an alternative means to implement adaptive coding. The basic premise of RCPC codes is to have a single encoder and decoder whose error correction capability can be modified by not transmitting certain coded bits (punturing the code). Puncturing is an effective adaptive transmission technique that is used in the IS-136, EDGE and HSDPA standards.

In this work, we consider an Adaptive Modulation and Coding (AMC) scheme where the power of the transmitted signal is held constant over the fixed frame interval, and the modulation and coding format is changed to match the current received signal quality or channel conditions. In a system with AMC, users with good channel conditions are typically assigned higher order modulation schemes with higher code rates (e.g. 64 QAM with R=3/4 turbo codes), but the modulation-order and/or code rate will decrease as the channel quality degrades (e.g. BPSK with R=1/3). Transmitting using a lower order modulation scheme and a smaller coding rate increases the reliability of transmissions due to a lower probability of error. AMC has been advocated at physical layer by many wireless network standards, such as 3GPP/3GPP2, HIPERLAN/2 and IEEE 802.11/16.

Several practical constraints determine when AMC can be used. Firstly, AMC requires a feedback path between the transmitter and the receiver which may not be feasible in some systems like GSM which are tailored for voice traffic. The use of AMC in there systems could lead to feedback delays which might adverely affect the transmission of voice traffic. Secondly, the effectiveness of adaptive techniques depends on the reliability of the estimated feedback from the receiver. If the channel changes faster than it can be reliably estimated, it can lead to poor performance. In addition, hardware constraints determine how often the transmitter can change its rate and/or power, which may limit performance gains acheivable through adaptive modulation. Finally, with AMC the transmission may be paused under poor channel conditions. This compromises the quality of fixed data rate applications like voice and video that have hard delay constraints.

1.2 Automatic Repeat Request

Another technique that is widely used to overcome the time-varying nature of the wireless medium is automatic repeat request (ARQ). ARQ is a link-layer technique that can increase the reliability of transmissions by retransmission of erroneously received packets. This technique improves the reliablility of transmission and provides for fine data rate adjustments when used with adaptive modulation and coding systems. The prerequisite for this method is a reverse channel between the transmitter and the receiver through which the success or failure of transmissions is conveyed to the transmitter as an acknowledgement.
There are two different aspects of retransmission techniques that are used to evaluate their performance [2] [3]. The first is the correctness of the technique that is a measure of the effectiveness of the protocol in sending packets successfully with the minimum number of retransmissions. The second aspect is efficiency. It measures how much of the transmitting capability of the channel is wasted by the unneccesary waiting and retransmissions.
We discuss the three main types of ARQ below.

Stop and Wait ARQ Stop and Wait ARQ (SW-ARQ) is the simplest ARQ technique. The basic premise is to ensure that each packet has been received correctly before transmitting the next packet. Thus, in transmitting packets from point A to B, the first packet is transmitted in the first frame, and the transmitter then waits for the acknowledgement(ACK) or negative acknowledgement (NACK) from B before transmitting the next packet. To ensure that the acknowledgements are received correctly, the ACK and NACK are protected with CRC.

The SW-ARQ technique is not commonly used in modern data networks because of its highly inefficient utilization of the communication channel. In particular, there is significant wastage of transmission bandwidth when the transmitter waits for ACK/NACKs from the receiver.

Go-Back-N ARQ Go back N ARQ (GBN-ARQ) is the most widely used type of ARQ protocol. In the GBN-ARQ protocol, all the packets in a frame, including and following the packet that was transmitted in error are retransmitted. This ensures a more or less continous data flow through the channel. With a transmit window that is unlimited in size, data packets with consecutive packet sequence numbers are transmitted until the transmitter receives a NACK for a packet that the receiver has recognized as being defective. Upon detection of an erroneous packet, the transmitter switches back to the defective packet based on the sequence number. The go back number $n \geq 1$, in a go back n protocol is a parameter that determines how many successive packets can be transmitted in the absence of a request for a new packet. Figure 1.2 illustrates the operation of go back 3 ARQ. Packet 1 was correctly received while packet 2 was received in error. The transmitter then starts to send packet 2 and the three subsequent packets that were in its buffer.

Selective-Repeat ARQ Go-back N ARQ, though more efficient than SW-ARQ still leads to sub-optimal link utilization while the transmitter waits for acknowledgements/ retransmissions. For some communication links with small error probabilities, the SR-ARQ technique proves to be the most efficient. The selective-repeat ARQ (SR-ARQ) protocol ensures continous transmission of packets over the channel. With this scheme, when an erroneous packet is detected, only the defective packet is selectively requested for retransmission. This increases the transmission efficiency at the cost of increased receiver complexity that arises due to the need for packet reordering.

Figure 1.3 shows an example of transmissions using SR-ARQ where the buffer size is $N = 3$. It can be seen the packet 2 is received in error, and the transmitter retransmits packet 2 as soon as it receives the NACK for that packet.

1.3 Adaptive Techniques in Current Systems

The mobile communication industry has experienced tremendous growth over the past decade beginning with the advent of the second generation cellular networks in the early 1990s. Unlike the first generation cellular systems that relied exclusively on frequency division multiple access (FDMA) and analog FM, the second generation standards use digital modulation formats and time division multiple access (TDMA). The advantages of digital communication systems with respect to signal regeneration and processing, use of error detection and correction, and better reliablility have led to the widespread adoption of the second generation mobile communication systems. Global System for Mobile Communication (GSM) and Code Division Multiple Access (CDMA) are two of the most successful second generation mobile communication standards. In this subsection, we discuss the adaptive techniques employed in these and emerging systems.

1.3.1 Second Generation Cellular Systems

GSM is the first mobile communication system using time-slotted digital transmission. It originally used two 25 MHz cellular bands - one for uplink and the other for downlink transmission. Frequency division duplexing (FDD), and a combination of time division multiple access (TDMA) and frequency hopping multiple access schemes are used to provide multiple access to mobile users. The available forward and reverse frequency bands are divided into 200 KHz wide channels labelled with different ARFCNs (Absolute Radio Frequency Channel Numbers). The Gaussian Minimum Shift Keying (GMSK) modulation technique is used for radio transmissions on both the forward and reverse link. The time slot number and an ARFCN constitutes a physical channel for both the forward and reverse links. Each physical channel can be mapped into different logical channels at different time slots. The GSM specification defines a wide range of logical channels that can be used to

Table 1.1. *GPRS Modulation and Coding Schemes*

Scheme	Code rate	Data rate (kbps)
CS-4	1	21.4
CS-3	3/4	15.6
CS-2	2/3	13.4
CS-1	1/2	9.05

link the physical layer with the data link layer of the GSM network. There are primarily two types of logical channels in GSM - the traffic channels (TCHs) and the control channels (CCHs). TCHs carry digitally encoded user speech or user data and have identical functions and formats on the forward and reverse links. CCHs are used to send the signalling and synchronization commands between the base station and the mobile terminal. There are three main control channels in the GSM system. These are the broadcast channel (BCH), the common control channel (CCCH), and the dedicated control channel (DCCH). The control channel consists of several logical channels that are distributed in time to provide the neccesary GSM control functions.

It can be seen from this discussion that the GSM standard provides limited mechanisms for adaptive transmission. This leads to sub-optimal spectral efficiency as the transmission strategy is independent of the fading channel conditions.

GSM was primarily designed for the efficient transmission of voice traffic using circuit switched techniques. In view of the inefficiency of circuit switched systems in supporting data transmission, General Packet Radio Service (GPRS) was introduced to provide packet data services to GSM-enabled mobile terminals. GPRS uses GMSK modulation similar to GSM. GPRS also uses adaptive coding and dynamic radio resource management to provide a maximum data rate of 171.2 kbps. Adaptive coding, which varies the transmission rate to the fading channel conditions, is the key radio interface improvement that differen-

Table 1.2. *EDGE Modulation and Coding Schemes (MCS)*

Scheme	Code rate	Modulation	Data rate (kbps)
MCS-9	1.0	8-PSK	59.2
MCS-8	0.92	8-PSK	54.4
MCS-7	0.76	8-PSK	44.8
MCS-6	0.49	8-PSK	29.6
MCS-5	0.37	8-PSK	22.4
MCS-4	1.0	8-PSK	17.6
MCS-3	0.80	GMSK	14.8
MCS-2	0.66	GMSK	11.2
MCS-1	0.53	GMSK	8.8

tiates GPRS from GSM. The coding techniques can be separated into three basic functions: channel coding, interleaving and puncturing. Channel encoding helps detect and correct bit errors. However, under good channel conditions, the error correction scheme may be an overkill. The transmitter may then use puncturing to remove some coded bits inorder to increase the user's data rate. Interleaving distributes the data across several time slots, thus making the receiver less susceptible to burst errors. GPRS uses various levels of puncturing and interleaving to yield a code rate that is appropriate for the channel characterisitics. Table 1.1 summarizes the coding schemes used for GPRS. The coding rate of the schemes are varied by puncturing the basic 1/2 rate convolutional code. The table lists the effective code rate and the raw data rate at the physical layer.

Enhanced Data Rate for Global Evolution (EDGE) is an enhancement to the GPRS radio technology to improve the throughput of packet data services. Adaptive modulation and coding together with varying puncturing rates are used for link adaptation. EDGE uses

8-phase shift keying (8-PSK) in addition to GMSK for adaptive modulation. Table 1.2 shows the nine different modulation and coding schemes (MCSs) for EDGE. The first column indicates the name of the MCS. The second column gives the effective code rate after puncturing. The two different modulation schemes that are used in combination with the different coding rates are given in the third column. The last column indicates the raw data rate of a single time slot using each MCS.

1.3.2 Third Generation Cellular Systems

Universal Mobile Telecommunication System (UMTS) is one of the major standards for Third Generation (3G) mobile communication systems. UMTS provides increased data rates of upto 2 Mbps per mobile thus addressing the growing demand for mobile Internet applications. UMTS was designed to provide a smooth transition from GSM networks by incorporating enhanced GSM Phase 2+ Core Networks with GPRS in its network architecture. UMTS makes use of wideband code division multiple access (W-CDMA) for more efficient air interface transmission. This is the primary difference between UMTS and previous GSM and GPRS networks. W-CDMA is a spread spectrum technique that enables multiple users to use the entire system bandwidth at the same time. This is accomplished by using orthogonal codes to separate transmissions. Each user is allocated its own unique spreading code. The information to be transmitted is encoded or spread for each data channel prior to transmission. The overall signal then assumes the rate of the code and hence its bandwidth. Thus, the transmitted signal or chip is wideband. The spread data channels are then added together and transmitted. The receiver recognizes the spreading code used for each data channel and can thus despread each individual data channel based on the received summation signal. UMTS offers limited means for adaptive transmission. Link adaptation is primarily acheived through fast power control. Fast power control ensures that each user transmits at just the power level that is required for successful reception. This reduces the interference level in a cell, and thereby increases system capacity.

High Speed Downlink Packet Acccess (HSDPA) is a 3GPP release 5 feature for UMTS,

Table 1.3. *HSDPA Modulation and Coding Schemes (MCS) [9]*

Scheme	Code rate	Modulation	Information rate (Mbps)
MCS-7	$3/4$	64	10.8
MCS-6	$3/4$	16	7.2
MCS-5	$1/2$	16	4.8
MCS-4	$3/4$	8	5.4
MCS-3	$3/4$	4	3.6
MCS-2	$1/2$	4	2.4
MCS-1	$1/4$	4	1.2

which was introduced to address the increasing demand for high data rate multimedia services. HSDPA provides key enhancements over W-CDMA release 99 for downlink transmission [4] [5]. It offers peak data rates of upto 10 Mbps resulting in better end-user experience for downlink data applications, with shorter connections and response times. In addition, HSDPA offers a three- to five-fold sector throughput increase, which supports a significantly larger number of users on a single frequency. The substantial increase in data rate and throughput is acheived by implementing joint AMC and Hybrid ARQ techniques. HSDPA uses QPSK as well as 16-QAM modulation schemes at the transmitter for adaptive transmission. In addition, coarse adjustment of the data rate is also acheived by varying the coding rate for transmissions. AMC is most effective when it is used in conjunction with fat pipe scheduling techniques [5] such as those enabled by the downlink shared channel. In addition to the benefits of fat-pipe scheduling, AMC combined with time domain scheduling offers the opportunity to take advantage of short term variations in a mobile user's fading envelope so that a user is always being served on a constructive fade.

Hybrid ARQ is an implicit link adaptation technique unlike AMC where explicit carrier/interference (C/I) measurements are used to set the modulation and coding format. In hybrid ARQ, link layer acknowledgements are used for re-transmission decisions. There are three types of hybrid ARQ that can be implemented - Chase combining, Rate compatible punctured turbo codes and Incremental redundancy [4]. Incremental redundancy (IR) or Type-II hybrid ARQ is another implementation of the hybrid ARQ technique wherein instead of sending simple repeats of the entire coded packet, additional redundant coding information is incrementally transmitted if the decoding fails on the first attempt. Type-III hybrid ARQ also belongs to the class of incremental redundancy ARQ schemes. However, with type-III hybrid ARQ, each retransmission is self-decodable which is not the case with type-II hybrid ARQ. Chase combining involves the retransmission by the transmitter of the same coded data packet. The decoder at the receiver combines these multiple copies of the transmitted packet weighted by the received SNR. Diversity(time) gain is thus obtained. In the type-III hybrid ARQ with multiple redundancy versions, different puncture bits are used in each retransmission. An enhanced version of the IR scheme called asynchronous and adaptive IR (A^2IR) is also proposed for HSDPA. The A^2IR scheme differs from Chase combining and IR in the sense that a different modulation and coding scheme can be selected for retransmissions than the original transmission. The standard IR scheme where the modulation and coding for retransmissions is the same as in the original is also referred to as non-adaptive IR scheme.

The choice of hybrid ARQ technique is an important design consideration. The two most commonly implemented ARQ schemes are selective repeat and SW-ARQ protocol. In SR-ARQ, only the blocks received in error are re-transmitted. The blocks are identified by sequence numbers. The SR-ARQ transmitter needs to transmit a number of blocks while awaiting a response. Hence, when combined with hybrid ARQ, the mobile needs to store soft samples for each partially transmitted block. Thus, the memory requirements can be huge. Besides, hybrid ARQ requires the receiver to know the sequence numbers of blocks before combining separate retransmissions. The sequence numbers have to be encoded separately with a higher coding rate to correct the errors induced by the channel conditions.

The increased redundancy results in an increase in the bandwidth required for signalling. SW-ARQ is a form of ARQ that requires very little overhead. In this technique, the transmitter operates on the present block until it has been received successfully. The protocol overhead is reduced by using a simple one-bit sequence number that identifies the current or next block. The acknowledgement overhead is reduced as the success or failure of decoding can be signalled using a single bit. Furthermore, the User Equipment (UE) memory requirements are minimized by having only a single block in transit at a time.
It can be seen that AMC provides for coarse adjustment in the data rate to adapt to the instantaneous channel conditions. These decisions are based on measurements either from UE measurement reports or network determined. For this mechanism to work efficiently, accurate measurements are required in addition to low feedback delay. An ARQ mechanism is required to supplement the AMC-based link adaptation scheme, as hybrid ARQ autonomously adapts to the instantaneous channel conditions, and is insensitive to measurement error and delay. Combining AMC with hybrid ARQ provides fine data rate adjustment of hybrid ARQ, in addition to the coarse data rate adjustment provided by AMC.

1.4 Related Work

Adaptive rate transmission was first propounded by Jeremiah F. Hayes, in 1968 [10]. Rate adaptation can be acheived through a variation of symbol time-duration or constellation size. The performance of an adaptive modulation system that varied the data rate by changing the pulse duration was examined in [11]. It was shown that significant improvements in power efficiency could be acheived by using adaptive transmission. However, this method requires complicated hardware and results in a variable bandwidth system, which is difficult to implement. Variation of the constellation size is better suited for hardware implementation since it results in a variable throughput system with a fixed bandwidth. This is done by selecting the modulation scheme based on channel state information that is fed-back from the receiver. Transmission is carried out at higher data rates by means of higher-order constellations when the channel conditions are conducive, and vice versa. The performance of adaptive modulations systems over Nakagami fading channels was studied in [12]. The authors derived the average bit error rate, spectral efficiency and outage probabality of the system in closed form. Later, adaptive rate transmission by varying the modulation scheme as well as the coding rate was proposed in [13]. In this work, the authors combined Trellis coding with adaptive modulation to significantly increase the spectral efficiency of adaptive modulation over fading channels.

Recently, adaptive modulation schemes have been considered together with automatic repeat request techniques to increase link reliablility. Conventionally, AMC and ARQ have been analyzed separately as they are techniques of different layers. Naijoh et al. [18] first showed through a simulation study that a joint AMC-ARQ transmission scheme can acheive high throughput in TDMA/TDD systems. The effect of type II hybrid-ARQ on adaptive modulation systems was examined in [19]. Later, some analytical expressions for the throughput of adaptive modulation systems combined with different ARQ strategies over fading channels were derived in [20] and [23]. The maximum delay of ARQ schemes can be reduced by using truncated ARQ to limit the number of retransmissions. The joint design of AMC with truncated ARQ was considered in [25], and later extended to MIMO systems using space-time block codes (STBC) in [24]. All the works on AMC and ARQ assume the continous availability of packets at the transmitter. However, in practice, queueing at the transmitter would be neccesary to acheive the desired spectral efficiency for AMC systems. Due to the use of finite-length buffers at the transmitter, packet loss and average delay of packets in the queue need to be taken into account while evaluating the performance of the system. Recently, Liu et al. examined the behaviour of the transmitter queue served by AMC- only transmission schemes [26] without considering the effect of retransmission.

The channel quality feedback (CQF) in AMC-Hybrid ARQ systems determine the downlink system performance. Frequent CQF results in good estimates of downlink channel quality at the base station, which, in turn, improves downlink system performance. However, this comes at the expense of larger uplink signaling overhead, thereby impacting the overall uplink capacity. Infrequent CQF, on the other hand, reduces this signaling overhead in the uplink at the expense of larger errors in the channel quality estimates available at the base station, thereby leading to system performance degradation. A variable CQF scheme that significantly reduces uplink signaling overhead without affecting downlink system performance, was examined in [15]. The performance comparision of different ARQ schemes being considered for HSDPA were evaluated in [14]. A simulation study was used to show that the A^2 IR scheme provides significant improvement in system throughput particularly at low mobility speeds. It was also shown that Hybrid ARQ (H-ARQ) gains at lower speeds are rather small compared to pure link adaptation with fast retransmissions but no combining. A cross-layer design for multiuser scheduling at the data link layer, with each user employing adaptive modulation and coding (AMC) was analyzed in [16]. The cross-layer scheduler guaranteed prescribed QoS for admitted QoS-guaranteed users and achieved efficient bandwidth utilization. Recently, this work has been extended to the joint design of scheduling, adaptive modulation and power control in wireless ad hoc networks in [17]. It was shown that considerable improvement in throughput and power efficiency could be gained by combining adaptive modulation with power control in adhoc wireless networks.

1.5 Significance of Research

The importance of AMC and ARQ techniques in improving the spectrum efficiency, throughput and link reliability of wireless systems has been discussed in the previous sections. The merits of combining these two techniques of the physical and link layers respectively, have led to the adoption of joint AMC and hybrid-ARQ schemes in the 3GPP specifications for High Speed Downlink Packet Access (HSDPA) systems [4].

The performance analysis of AMC-ARQ systems in existing literature have not taken into account the finite-length buffer constraints at the transmitter. A joint AMC-ARQ transmission scheme necessitates the use of a buffer at the transmitter. For a fixed modulation scheme with a constant data-rate of tranmsission, the number of packets leaving the buffer during a frame interval would be constant. However, this changes when an AMC-ARQ transmission scheme is adopted. The number of packets that leave the buffer now depends on the AMC mode that is chosen for transmission, the success or failure of previous transmissions, and the retransmission technique that is adopted. We consider a finite-length buffer at the transmitter as in all practical systems. Finite-length buffering can lead to packet losses due to buffer overflow, and to increased average wait-times for packets in the queue. Therefore, analysis of AMC-ARQ systems with finite length buffers is particularly important.

In this thesis, we design and analyze more integrated joint AMC-ARQ systems. We take into account the practical constraint of the finite-length transmit buffer in our design. For the resulting systems, we carry out a thorough performance and efficiency analysis based on Markov chain theory. Further, a new AMC strategy is developed that takes into account the channel conditions as well as the state of the buffer while choosing the mode of transmission. The resulting improvement in reliability of transmissions is shown using Markov chain analysis. The results presented in this thesis will provide valuable guidelines in the design of the next generation of wireless systems.

1.6 Thesis Outline

The remaining of the thesis is organized as follows. Chapter 2 introduces the joint AMC-ARQ transmission system. In Chapter 3, we analyze the performance of SNR-based joint AMC-ARQ systems using Markov chain theory. The performance analysis of the buffer-assisted, joint AMC-ARQ system is presented in Chapter 4. The conclusions drawn from the work are then discussed in Chapter 5.

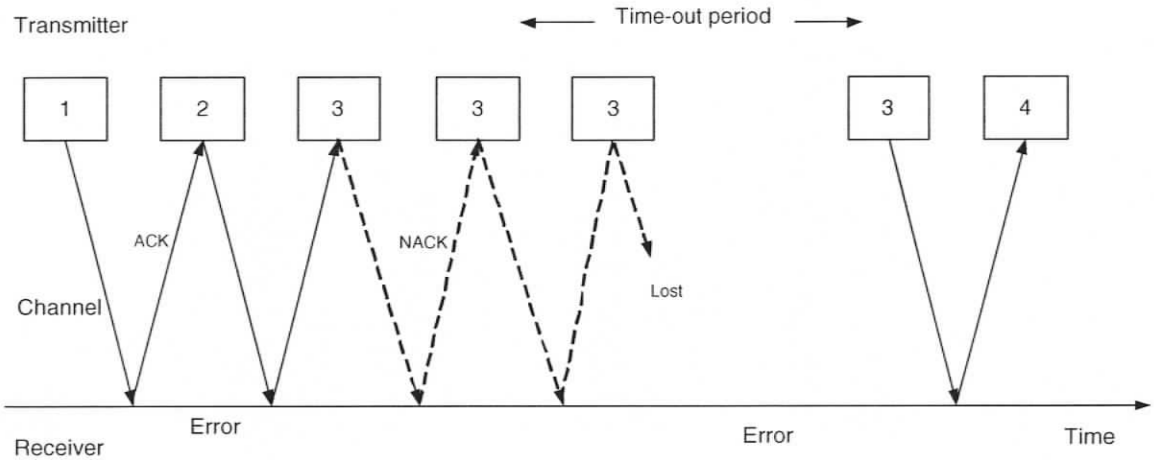

Figure 1.1. *Stop and Wait ARQ.*

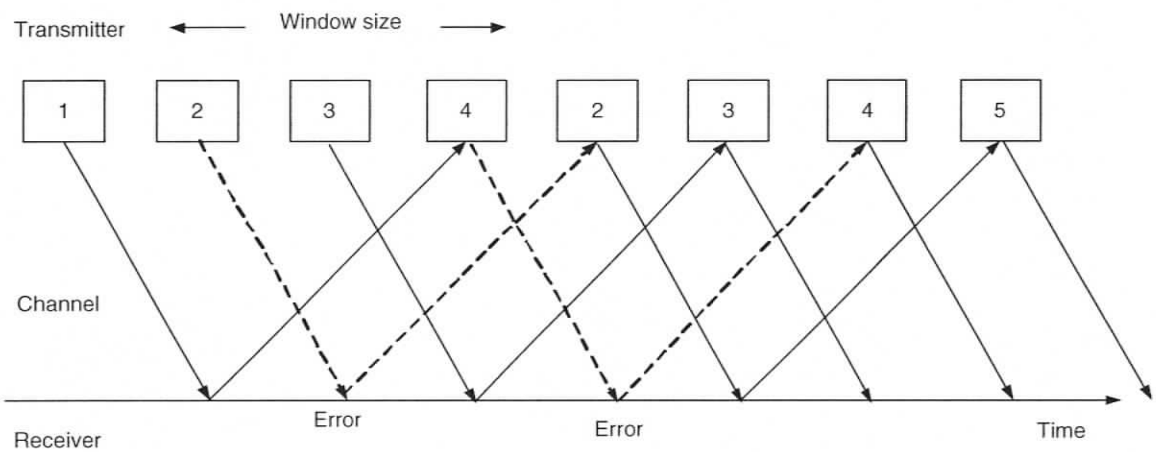

Figure 1.2. *Go-Back N ARQ.*

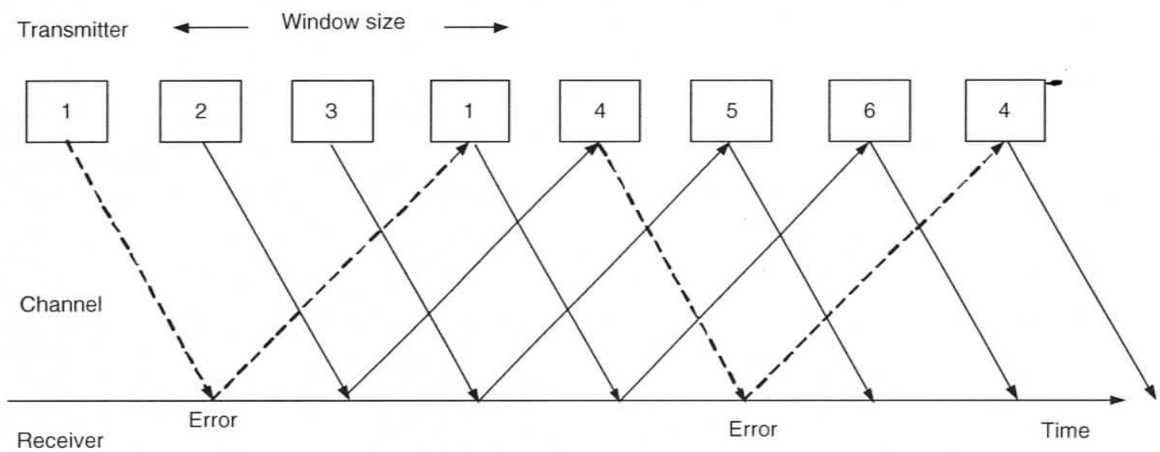

Figure 1.3. *SR-ARQ.*

Chapter 2

Joint AMC-ARQ Transmission System

The framework for the joint AMC-ARQ transmission system is described in this chapter. We first present the system structure in section 2.1. The AMC model that is adopted in this work is presented in section 2.2. Further, the closed form expression for the packet error rate of AMC systems is presented as well. In section 2.3, we discuss the operation of the joint AMC-ARQ system. We then describe the traffic and channel models in section 2.4.

2.1 System Structure

A TDMAbased medium access scheme that has a frame structure as shown in Fig. 2.1 is considered. The processing unit at the data link layer is a packet containing multiple information bits, and the processing unit at the physical layer is a frame [25]. The frame consists of multiple time slots for different users. The length of each user slot is T_s while that of the whole frame is T_f . Therefore, transmissions for a particular active user happens every T_f time period for a duration of T_s. Each user transmits using the joint AMC-ARQ transmission scheme during the allocated time slot. Both the AMC and ARQ feedback occurs during the guard interval that is used to separate frames.

Fig. 2.2 illustrates the system model for the joint AMC-ARQ transmission. The AMC controller at the receiver determines the combination of modulation and coding schemes based on the estimated channel state information. This combination of modulation and coding scheme is defined as a mode for transmission. The mode selection is then fedback to the transmitter where the AMC controller of the transmitter determines the number of packets that are to be transmitted based on the mode selection and the number of packets in the queue. The ARQ controller at the receiver determines the packets that have to be retransmitted, and sends this information to the ARQ controller at the transmitter through a feedback channel. The ARQ controller at the transmitter then removes the appropriate packets from the buffer for retransmission. Both the feedback channels are assumed to be error free. An error feedback channel is made possible by making use of powerful error control coding.

2.2 Adaptive Modulation and Coding

A constant power and variable rate AMC scheme with five transmission modes ($M = 5$) is considered, as shown in Table 1. Here, a_i and g_i are parameters that depend on the chosen modulation and coding scheme. The total range of received SNR is partitioned into $M + 1$ nonoverlapping intervals. The boundary SNRs are denoted by $\{\gamma\}_{i=0}^{m+1}$. The boundaries for the AMC scheme are chosen so that it is the mininum SNR required to acheive a target instantaneous packet error rate P_{target}. The SNR thresholds can be determined as:

$$\begin{aligned} \gamma_0 &= 0, \\ \gamma_n &= \frac{1}{g_n} \ln\left(\frac{a_n}{P_{target}}\right), \quad n = 1, 2, ...M + 1, \\ \gamma_N &= +\infty, \end{aligned} \tag{2.1}$$

With the traditional AMC scheme, when the estimated SNR falls in the interval (γ_i, γ_{i+1}), then the AMC mode i is chosen for transmission. Mode 0 corresponds to the case where no packet is transmitted. The exact packet error rate for an adaptive modulation system with convolutional coding cannot be obtained in closed form. However, an approximation for the instantaneous packet error rate of an AMC system has been derived as: [25]:

$$per_i(\gamma) \approx \begin{cases} 1 & \text{if } 0 < \gamma < \gamma_i \\ a_i \exp(-g_i\gamma) & \text{if } \gamma \geq \gamma_i, \end{cases} \tag{2.2}$$

Table 2.1. *AMC look-up table*

	Mode 1	**Mode 2**	**Mode 3**	**Mode 4**	**Mode 5**
Modulation	BPSK	QPSK	QPSK	16-QAM	16-QAM
Coding rate R_c	$1/2$	$1/2$	$3/4$	$9/16$	$3/4$
Rate (bits/symbol)	0.50	1.00	1.50	2.25	3.00
a_i	274.72	90.25	67.61	50.12	53.39
g_i	7.99	3.49	1.68	0.66	0.37
γ_i	-1.5331	1.0942	3.9722	7.7021	10.2488

2.3 Joint AMC-ARQ Transmission

In traditional TDMA systems, a fixed modulation scheme such as DQPSK or GMSK, is used for packet transmission over each time slot. With the AMC-ARQ system, both the modulation scheme used and the number of packets transmitted over the time slot varies. In this section, we discuss the operation of the proposed AMC-ARQ system in detail.

In the proposed system, retransmission is used to improve link reliability. In particular, whenever there is a transmission error during the current time-slot, the receiver informs the transmitter by feeding back the nature of errors before the next transmission slot. The transmitter will then retransmit certain packets as per the rules given below. We assume that the error detection scheme used at the receiver is perfect. In addition, we assume that both the AMC and ARQ feedback occurs within a frame interval T_f, i.e. during every guard interval. Feedback delay can affect the performance of AMC as the delay can result in decoding errors at the receiver. Erroneous or delayed feedback can also significantly influence the performance of retransmission schemes. The absence of an acknowledgement due to transmission errors is generally treated as a negative ACK. Hence, the effect is similar to that of an error in the forward channel. The performance of retransmission schemes under delayed feedback conditions has been analyzed in [21] and [22]. These works have examined queueing with ARQ schemes that have a feedback delay. However, in this thesis we assume negligible feedback delay to keep the analysis of the joint AMC-ARQ system tractable. The analysis of joint AMC-ARQ systems with feedback delay is a problem worth considering in future work.

A finite length buffer at the transmitter is implemented to account for the occasional packet retransmission. Packets are removed from the buffer only when they have been succesfully received. Therefore, packet loss can occur in the system only when a new packet arrives to find that the buffer is full.Depending on the selected AMC mode, variable number of information symbols can be transmitted during a fixed slot duration of T_s. We assume that T_s is chosen such that the lowest AMC transmitting mode, i.e. Mode 1, allows for the transmission of exactly one packet over one transmission slot. It can be noted from Table I, that the rate of AMC Mode i is greater than or equal to i times the rate of Mode 1. As such, we assume that a maximum of i packets can be transmitted over one time slot using Mode i.

We consider two AMC mode selection strategies. With the first strategy termed, as SNR-based AMC, the mode selection is solely based on SNR estimation at the receiver. In Fig 2.3 , we illustrate the operation of the SNR-based AMC controller through two examples. During the first time slot, the receiver selects AMC mode 5, which allows five packets to be transmitted during the time slot. However, there are only three packets in the buffer, hence three packets are transmitted during the first time slot. Similarly, it can be seen that two packets are tranmitted during the second time slot as the channel conditions support the choice of AMC mode 2.

In the second strategy, termed buffer-assisted AMC, the mode of transmission is selected based on the fading channel conditions as well as the state of the finite-length buffer at the transmitter. More specifically, the transmitter will select the lowest AMC mode that the fading channel condition permits so that packets are tranmsitted with a lesser probability of error. Thus the likelihood of selecting a higher order mode than is required for transmission of all packet remaining in the buffer, is reduced to zero. Consider the examples shown in Fig. (2.4). With the buffer-assisted AMC scheme, the transmitter selects AMC mode 3 for transmission during the first transmission slot even though the channel conditions support the use of transmission mode 5. This is due to the number of packets in the buffer being less than the mode of transmission that is supported by fading channel conditions. The mode selection strategy is acheived by asking the transmitter to send a mode selection update based on the buffer status to the receiver before the second time slot starts over a certain forward channel. It can be seen that the use of the buffer-assisted AMC-ARQ transmission scheme leads to the choice of the mode 3 as opposed to mode 5 by the SNR-based AMC-ARQ, under similar channel conditions. This ensures that the buffer-assisted AMC-ARQ systems transmit using the lowest possible mode required for successful tranmission of packets in the buffer. During the second time interval, the transmitter selects AMC mode 2, which is the highest mode of transmission supported by the prevailing channel conditions.

Since multiple packets are transmitted over one time slot, several retransmission strategies apply. We evaluate three buffer management strategies. All retransmit (AR) is a basic buffer management scheme wherein all packets are retransmitted in the next time slot, if at least one packet in the present frame is received erroneously in the current slot. This is applicable to scenarios where all packets during a time slot are jointly encoded. On the other hand, when the packets are independently encoded, we can improve the transmission efficiency using two buffer management strategies, Partial retransmit (PR) and Selective retransmit (SR). In PR, the transmitter will retransmit the first erroneous packet and all following packets in a frame. It can be seen that the PR strategy maintains the packet order during transmission. With SR, only those packets that are received in error will be retransmitted. This increases the system complexity as packet re-ordering will be neccesary at the receiver.

2.4 Traffic and Channel Model

In the following two chapters, we will analyze the two mode selection strategies with different buffer management strategies. In the analysis, we assume the following traffic model. Two types of traffic arrival patterns are considered- steady stream for real time traffic, and the generalized Poisson traffic to represent traffic with varying traffic intensity. The Poisson traffic arrival pattern also includes steady stream model as a special case, when the average arrival rate equals the peak arrival rate. A packet containing multiple information symbols is the basic information unit. For steady stream traffic, one and only one packet arrives at the transmitter during one frame duration, whereas for realistic Poisson traffic, the number of packets that arrive during one frame duration is random and varies from zero to a certain maximum value. A slowly varying flat-fading channel is considered in this study, so that the channel changes at a rate much slower than the symbol data rate. This implies that the channel remains constant over hundreds of symbols. The Nakagami block fading channel model can be used to represent both severe and weak fading. In addition, it includes the Rayleigh fading model as a special case. Finally, it gives the best fit for urban and indoor multipath propagation [25]. Considering all the aforementioned factors, we use the Nakagami block fading channel model. The fading experienced by a user during successive frame durations is assumed to be uncorrelated. The probability density function (pdf) of received SNR is given by:

$$p_\gamma(\gamma) = \frac{m^m \gamma^{m-1}}{\overline{\gamma}^m \, \Gamma(m)} \, exp\left(\frac{-m\gamma}{\overline{\gamma}}\right), \tag{2.3}$$

where $\overline{\gamma} = \mathrm{E}\left\{\gamma\right\}$ is the average value of the received SNR, $\Gamma(m)$ is the Gamma function, and m is the Nakagami fading parameter. Correspondingly, the average packet-error rate ($\overline{PER_i}$) while using the ith transmission mode can then be defined as follows [25]:

$$\begin{aligned} \overline{PER_i} &= \frac{1}{P_i}\int_{\gamma_i}^{\gamma_{i+1}} per_i(\gamma).\, p_\gamma(\gamma) d\gamma \\ &= \frac{1}{P_i}\frac{a_i}{\Gamma(m)}\left(\frac{m}{\overline{\gamma}}\right)^m \times \frac{\Gamma(m, b_i\gamma_i) - \Gamma(m, b_i\gamma_{i+1})}{(b_i)^m}, \end{aligned} \tag{2.4}$$

where $b_i := m/\hat{\gamma}+g_i$, and $\Gamma(m, x)$ is the Gamma function. P_i is the probability of choosing a mode for transmission, which is given by [25]:

$$\begin{aligned} P_i &= \int_{\gamma_i}^{\gamma_{i+1}} p_\gamma(\gamma) d\gamma \\ &= \frac{\Gamma\left(m, \frac{m\gamma_i}{\hat{\gamma}}\right) - \Gamma\left(m, \frac{m\gamma_{i+1}}{\hat{\gamma}}\right)}{\Gamma\left(m\right)}. \end{aligned} \tag{2.5}$$

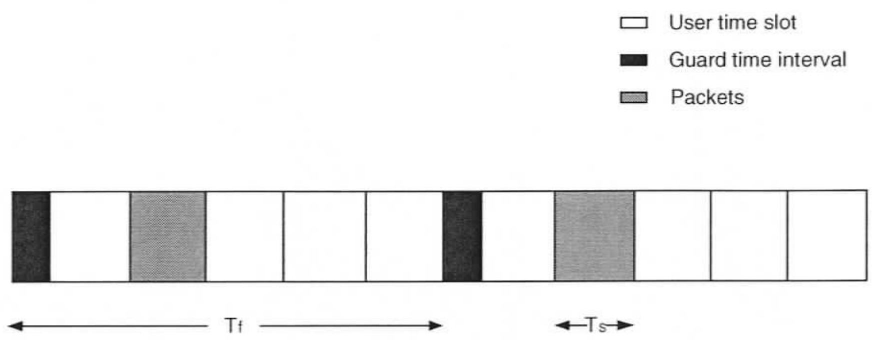

Figure 2.1. *Frame structure*

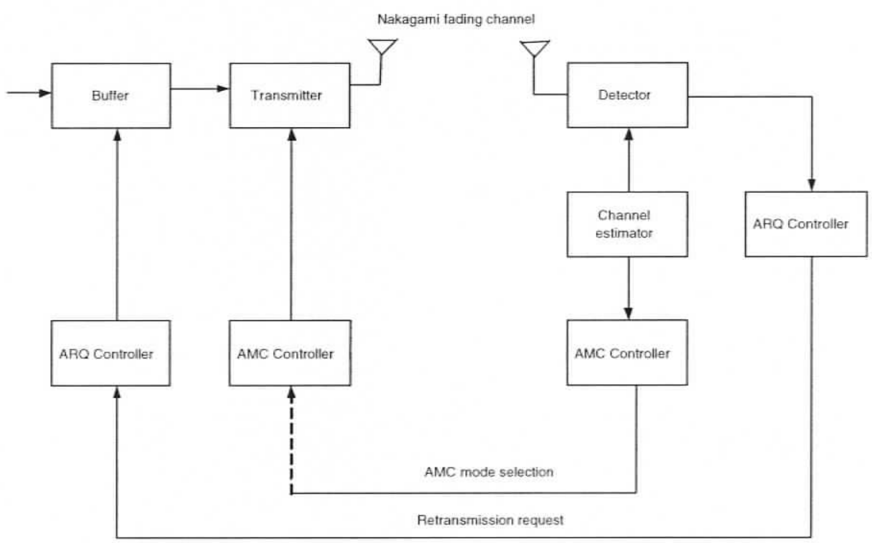

Figure 2.2. *System model.*

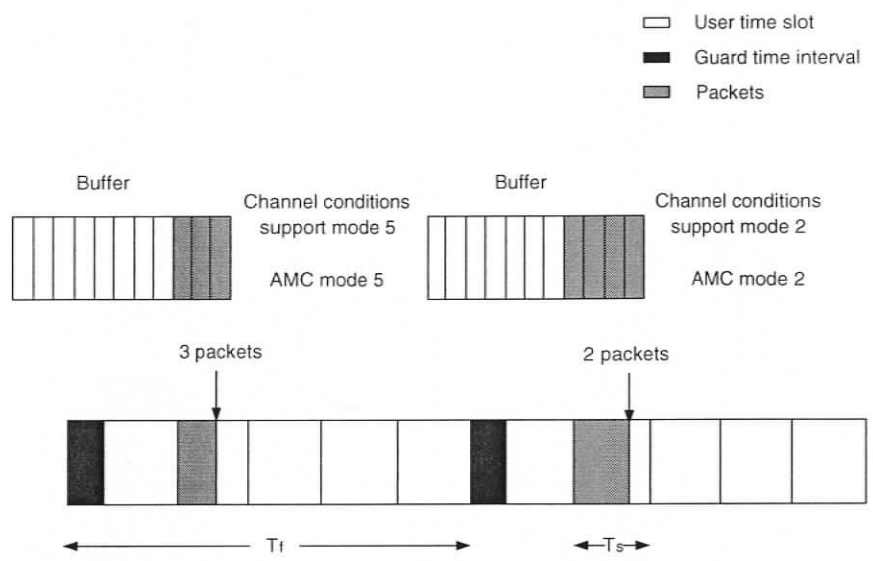

Figure 2.3. *Frame structure for AMC-ARQ transmission.*

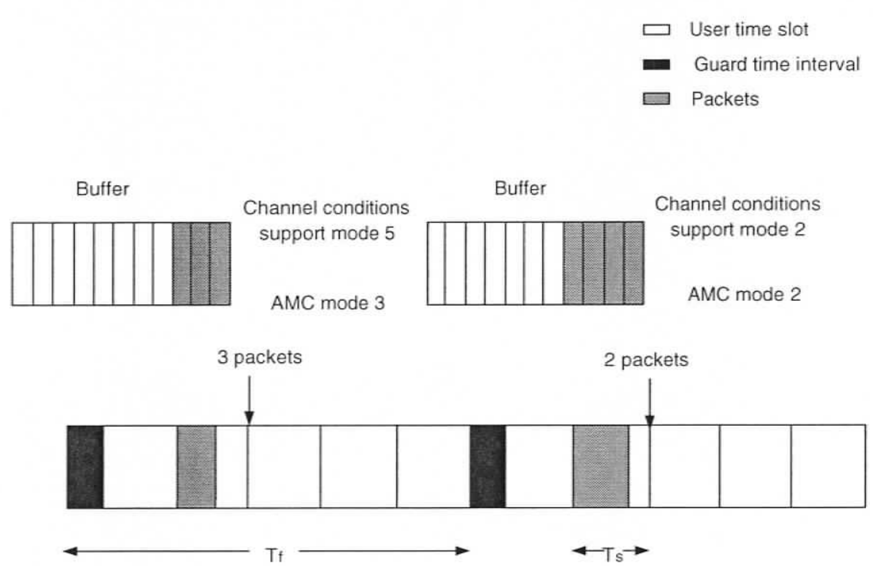

Figure 2.4. *Frame structure for buffer-assisted AMC*

Chapter 3

Analysis of SNR-based Joint AMC-ARQ Transmission

In this chapter, we develop a Markov chain model for the performance analysis of joint AMC-ARQ transmission systems. We develop the Markov chain model in section 3.1, which is then used to study the performance of the SNR-based joint AMC-ARQ transmission system for the steady stream traffic case. In section 3.2, the Markov model is extended to analyze the performance of the SNR-based joint AMC-ARQ transmission system for the realistic Poisson traffic case. The performance is measured in terms of the packet loss rates, average wait time and spectral efficiency of the systems.

3.1 Steady Stream Traffic Case

We consider the transmission of steady-stream traffic in this section. Note that with steady stream traffic, one and only one packet arrives at the transmitter during one frame duration.

3.1.1 Markov Chain Model

We develop a Markov chain model to characterize the behaviour of the transmitter buffer. This framework is then used to analyze the proposed SNR-based joint AMC-ARQ system in the following sections. We define the state space to be the number of packets in the buffer at the begining of each time slot. For steady state traffic, a packet arrives at the buffer during each frame interval. Therefore, for the purpose of analysis, a packet can be assumed to arrive at the begining of each transmission slot. The buffer will then contain atleast one packet at the begining of each transmission slot. The size of the state space is equal to the buffer length, denoted by B and state n, $1 \leq n \leq B$, corresponds to the event that there are n packets in the buffer at the begining of a time slot. Let p_{uv} denote the transition probability from state v to state u where $1 \leq u, v \leq B$. Thus the transition probability matrix for the buffer can be expressed in the form:

$$P = \begin{pmatrix} p_{11} & p_{12} & \cdots & p_{1B} \\ p_{21} & p_{22} & \cdots & p_{2B} \\ \vdots & \vdots & \ddots & \vdots \\ p_{B1} & p_{B2} & \cdots & p_{BB} \end{pmatrix}_{B \times B}$$

Note that p_{uv} is also the probability that the number of packets in the buffer changes from v to u over one time slot. These probabilities can be determined based on the dynamics of the arrival and transmission processes. For the steady stream process under consideration, where a new packet arrives every T_p time duration with probability one, we can see that $p_{uv} = 0$ if (i) $u > v + 1$ because the buffer queue size can increase by atmost one over one time slot; or (ii) $u < v - M$ because the queue size can decrease by atmost M over one time slot. The queue can be full during succesive time slots only if either no packets or only one packet leaves the queue. For other cases, i.e. $v + 1 \geq u \geq v - M$, $p_{u,v}$ is equal to the probability that $v - u + 1$ packets leave the queue given that there were v packets in the queue. Mathematically speaking, we have:

$$p_{u,v} = \begin{cases} 0 & \text{if } u < v - M \text{ or } u > v + 1 \\ z_{0,B} + z_{1,B} & \text{if } u = v = B \\ z_{v-u+1,v} & \text{otherwise,} \end{cases} \tag{3.1}$$

where $z_{k,s}$, $0 \leq k \leq \min[s, M]$, $1 \leq s \leq B$, denotes the probability that k packets leave the queue given that there were s packets in the queue. The probabilities $z_{k,s}$ are calculated for the different buffer management strategies in the following.

3.1.1.1 AR Scheme

In the AR scheme, all packets in a frame are retransmitted if any of the packets are received in error during the current time slot. As such, either all $k > 0$ transmitted packets leave the queue or no packet leaves the queue. In particular, $k > 0$ packets leave the queue given that there are s packets in the queue if (i) Mode k is chosen and all packets are received correctly for $0 < k < \min[s, M]$ or (ii) one of the modes $k, k+1, .., M$ is chosen and all packets are received correctly for $k = s \leq M$. No packet leaves the queue given that there are s packets in the queue if (i) one of the M possible transmitting modes are chosen and there is atleast one packet detected in error or (ii) Mode 0 is chosen. Therefore, $z_{k,s} \leq \min[s, M]$ for the AR scheme can be calculated as:

$$z_{k,s} = \begin{cases} \sum_{i=k}^{M} P_i \, (1 - \overline{PER_i})^k; & k = s \leq M \\ P_k \, (1 - \overline{PER_k})^k & 0 < k < min[s, M+1] \\ P_0 + \sum_{i=1}^{M} P_i \, [1 - (1 - \overline{PER_i})^{\min(s,i)}], & k = 0 \end{cases} \quad (3.2)$$

where $\overline{PER_i}$ is the average packet error rate for the ith mode and P_i is the probability of choosing AMC Mode i, which was given in (3) and (4) respectively.

3.1.1.2 PR Scheme

With the PR scheme, a packet received in error is retransmitted along with all subsequent packets in the frame, during the next time slot. In this case, k packets leave the queue if and only if the first k packets transmitted are correctly received and the $k+1$th packet, if transmitted, is detected in error. The probability $z_{k,s}$ for the PR scheme can be shown to be given by:

$$z_{k,s} = \begin{cases} \sum_{i=k}^{M} P_i \left(1 - \overline{PER_i}\right)^k & k = s \leq M \\ P_k \left(1 - \overline{PER_k}\right)^k & \\ + \sum_{i=k+1}^{M} P_i \left(1 - \overline{PER_i}\right)^k \overline{PER_i} & 0 \leq k < min[s, M+1]. \end{cases} \tag{3.3}$$

3.1.1.3 SR Scheme

With the SR scheme, only packets received in error during the current time slot are to be retransmitted. As one would intuitively expect, the transmission efficiency of this scheme is higher than both AR and PR schemes. However, packets may arrive out of order, and packet reordering introduces additional complexity at the receiver. It can be shown that the probability $z_{k,s}$ for the SR scheme is given by:

$$z_{k,s} = \delta(k) P_0 + \sum_{i=k}^{M} P_i \begin{pmatrix} \min[s,i] \\ k \end{pmatrix} (1 - \overline{PER_i})^k (\overline{PER_i})^{\min[s,i]-k}, \tag{3.4}$$

where $\delta(.)$ is the Delta function.

3.1.2 Performance Analysis

We now apply the transition probabilities of the Markov chain for the three buffer management schemes to study the performance of the proposed AMC-ARQ system. We first derive the steady state behavior of the Markov chain. Let $\pi = [\pi_1, \pi_2 .... \pi_B]$ denote the steady state distribution vector, which can be obtained as the normalized eigenvector of the matrix P corresponding to the eigenvalue of unity. For a particular choice of buffer size B, the steady state distribution vector can be calculated analytically in terms of $z_{k,s}$. For example, it can be shown that for $B = 3$ and $M > B$, the stationary vector π is given by:

$$\begin{aligned}\pi = \Big[& \frac{\eta_3}{\eta_3 + z_{0,1}z_{0,2} + z_{0,1}(1 - z_{1,3})}, \\ & \frac{z_{0,1}(1 - z_{1,3})}{\eta_3 + z_{0,1}z_{0,2} + z_{0,1}(1 - z_{1,3})}, \\ & \frac{z_{0,1}z_{0,2}}{\eta_3 + z_{0,1}z_{0,2} + z_{0,1}(1 - z_{1,3})}\Big],\end{aligned} \tag{3.5}$$

where $\eta_3 = (1-z_{1,2})(1-z_{1,3})-z_{2,3}z_{0,2}$. We now evaluate the packet loss rate, average delay and transmission efficiency of the proposed channel-adaptive transmission/retransmission technique for steady-stream traffic in the following. The analytical results are validated using simulations. The simulations were carried out in MATLAB with 10^7 iterations for each plot.

3.1.2.1 Packet Loss Rate

For the transmission of steady stream traffic using the proposed transmission system, packet loss occurs only if a packet arrives at the beginning of a time slot and finds that the buffer is full. Thus, the packet loss probability can be calculated as the probability that the queue is full, times the probability that no packet leaves the queue over the next time slot, given that the queue is full i.e.

$$P_{Loss} = \pi_B z_{0,B}. \tag{3.6}$$

For B=3, the packet loss can be expressed as follows using equation 3.5:

$$P_{loss} = \frac{z_{0,1}z_{0,2}z_{0,3}}{\eta_3 + z_{0,1}z_{0,2} + z_{0,1}(1 - z_{1,3})}, \tag{3.7}$$

where η_3 was given above. Fig. 3.1 illustrates the packet loss performance of the proposed transmission system for steady stream traffic. For reference, the packet loss rate of a traditional system with fixed BPSK modulation is also plotted. It can be seen that even with a small buffer size of $B = 2$, the proposed system can offer considerable performance improvement over traditional systems, and this improvement increases with an increase in buffer size. As expected, for a fixed buffer size, SR strategy leads to the smallest packet loss rate while AR leads to the largest loss rate. Note that the performance advantage of SR comes at the cost of additional complexity of packet reordering.

3.1.2.2 Average Delay

Average queueing delay of a packet is defined as the average number of time slots that a packet stays in the buffer before being correctly received. It is assumed that if a packet is successfully received during its initial transmission, then its wait time is equal to one time slot, which is the minimum. Direct calculation of the average number of time slots requires the derivation of the probability mass function, which is quite complex and differs between different buffer management strategies. Using Little's law on the buffer subsystem [2] , the average number of time slots that a packet needs to wait can be calculated for all three buffer management strategies as:

$$W = Q_a/\lambda, \tag{3.8}$$

where λ is the arrival rate of the buffer, which is equal to 1 for steady stream traffic, and Q_a is the average queue length that can be determined using the stationary probability s_i as:

$$Q_a = \sum_{i=1}^{B} i\pi_i. \tag{3.9}$$

The average packet delay for $B = 3$, after applying (3.5) and some manipulation, can be shown to be:

$$W = \frac{\eta_3 + 3z_{0,1}z_{0,2} + 2z_{0,1}(1 - z_{1,3})}{\eta_3 + z_{0,1}z_{0,2} + z_{0,1}(1 - z_{1,3})}, \tag{3.10}$$

where η_3 was given in (3.1.2).

The delay performance is measured in terms of the average delay of a packet in the queue, which is the number of time slots that a packet has to wait before being transmitted. Figure 3.2 shows the delay performance of the proposed system for steady stream traffic. It can be seen that the delay performance of the three retransmit strategies are very similar. When the average received SNR increases from -5 to 25dB, the average delay of the three strategies decreases from B to 1, which means that the packets are immediately transmitted successfully to the receiver over the subsequent time slot. It can be concluded that the strategies with higher transmission efficiency, e.g. SR and PR, will lead to a lower packet loss rate but not a smaller average delay. Comparing the curves corresponding to the same transmission strategy, we observe that improved packet loss performance with increased buffer size comes at the cost of increased delay over the low SNR region. It should also be noted that since for AR and PR strategies, a packet will stay in the buffer for atmost B time slots, the delay of the proposed system is bounded. Therefore, the design parameter B for these strategies can be determined based on the maximum delay constraint.

3.1.2.3 Transmission Efficiency

We now examine the efficiency of the proposed system. In particular, we focus on the spectral efficiency and power efficiency of the system. With the assumption of equal-length packets, the average number of packets transmitted during a time slot serves as a good measure of the physical layer spectral efficiency. Note that if the received SNR falls into the ith interval and the buffer has v packets for transmission, the transmitter will transmit $\min[i, v]$ packets during the next transmission slot. Therefore, the physical layer spectral efficiency, denoted by η_{PHY} is given by:

$$\eta_{PHY} = \sum_{i=1}^{M} P_i \sum_{v=1}^{B} \pi_v \min[i, v], \tag{3.11}$$

where P_i is the probability of selecting Mode i and π_v is the steady state probability of state v, both of which are given before. Note that the performance improvement with the proposed transmission system comes at the cost of occasional retransmission of certain packets. In the following, we evaluate its power efficiency by calculating the average number of transmission/retransmissions that the transmitter needs to perform for each successfully received packet, denoted by N_t. Note that for steady stream traffic under consideration, the average number of packets being succesfully received over one time slot is equal to $1 - P_{Loss}$. N_t can be calculated as:

$$N_t = \frac{\eta_{PHY}}{1 - P_{Loss}} = \frac{1}{1 - \pi_B z_{0,B}} \sum_{i=1}^{M} P_i \sum_{v=1}^{B} \pi_v \min[i, v], \tag{3.12}$$

We illustrate the transmission efficiency of the proposed system for steady stream traffic in Fig.3.3. Fig.3.3(a) plots the average number of packets transmitted during a time slot for the three strategies. It can be seen that when the average received SNR increases, the spectral efficiency improves and saturates to the maximum of one packet per time slot, as expected. Similar to the delay performance, the spectral efficiency of the three strategies are identical. The average number of packet transmissions per succesive time slot for the three strategies are plotted in Fig 3.3(b). It can be observed that AR needs a larger number of retransmissions for successfully delivering one packet and therefore has the lowest power efficiency. It can also been seen that as the average SNR become too very large or small, the average number of packet transmission per successive packet reduces to one. Intutively, when the channel condition is good, all packets will be successfully delivered upon the initial transmission. On the other hand, when the channel condition is poor, no transmission attempt will be made and most packets will be dropped due to buffer overflow.

3.2 Realistic Poission Traffic Case

In this section, we evaluate the performance of the proposed system when the traffic arrival pattern follows the realistic Poisson model [2]. The realistic Poisson traffic model proposed in [2] describes traffic that can have varying intensities more accurately than the normal Poission model. The model is characterized by two parameters: average packet arrival rate in terms of the maximum number of packet arrivals over one time interval, denoted by λ, and peak arrival rate in terms of the maximum number of packet arrivals over one time interval, denoted by N_m. With this arrival model, it can be shown that the probability that k packets arrive over one time interval is given by:

$$q(k) = \begin{pmatrix} \mathrm{N}_m \\ k \end{pmatrix} \Big(\frac{\lambda}{N_m}\Big)^k \Big(1 - \frac{\lambda}{N_m}\Big)^{N_m - k}, \quad 0 \leq k \leq N_m. \tag{3.13}$$

Note that when $N_m = \lambda$, we have $q_k = 1$ if $k = N_m$ and 0 otherwise. The realistic Poisson traffic arrival pattern includes the general steady stream traffic, where exactly λ packets arrive during one time interval with probability one, as special cases.

3.2.1 Markov Chain Model

We now develop a Markov chain model for the realistic Poisson traffic case. For Poisson traffic arrival, the probability that no packets arrive at the queue is not zero. Hence, the queue size can be zero at the begining of the time slot. We increase the size of the state space of the Markov chain by one to account for the case of no packet in the queue. In particular the queue can be modeled as a $B+1$ state Markov chain with state n, $0 \leq n \leq B$, corresponding to the case that there are n packets in the queue at the begining of the transmission slot. The transition probability for the $B+1$ state Markov chain is given by:

$$P = \begin{pmatrix} p_{00} & p_{01} & \cdots & p_{0B} \\ p_{10} & p_{11} & \cdots & p_{1B} \\ \vdots & \vdots & \ddots & \vdots \\ p_{B0} & p_{B1} & \cdots & p_{BB} \end{pmatrix}_{B+1 \times B+1}$$

The transition probability is arrived at separately for the two cases: $u < B$ and $u = B$. The probability that the buffer size changes from v to u, where $u < B$ over one time step is equal to the probability that n packets are removed given that there are v packets, and $u - (v - n)$ packets arrive over one time step, where $n \leq j$ and $0 \leq u - (v - n) \leq N_m$. Assuming that the arrival and departure processes are independent, it can be shown that the transition probability p_{uv} is given by:

$$p_{uv} = \sum_{n=\max[0, v-u]}^{\min[v, N_m - (u-v)]} q_{u-(v-n)} z_{n,v}, \quad 0 \leq u < B, \tag{3.14}$$

where q_k was given in (3.13) and $z_{k,s}$ was derived for different buffer management strategies in the previous section. We also need to take into account the case where the buffer is full ($u = B$), and more packets arrive than the buffer can accomodate. Then, the excess packets will be dropped. The probability $p_{B,v}$ is equal to the probability that n packets leave the queue given that there are v packets, and $B-(v-n)$ or more packets arrive over one time step, where $n \leq v$ and $0 \leq B-(v-n) \leq N_m$. Mathematically speaking, we have:

$$p_{uv} = \sum_{n=0}^{\min[v, N_m-(B-v)]} \Big(\sum_{k=B-(v-n)}^{N_m} q_k \Big) z_{n,v}, \quad i = B. \tag{3.15}$$

By setting $N_m = \lambda$ in the above two equations, we can obtain the transition probabilities for the general steady stream traffic case where exactly λ packets arrive over each time interval.

3.2.2 Performance Analysis

The performance of the proposed transmission system is now analysed for realistic Poisson traffic arrival over Nakagami fading channel conditions. We again focus on average packet loss rate, average delay and transmission efficiency statistics.

3.2.2.1 Packet loss rate

For the realistic Poisson traffic under consideration, the average packet loss rate can be calculated as the ratio of average packet loss over one time step, denoted by N_L, and average number of packet arrivals over one time step, λ i.e. $P_{loss} = N_L/\lambda$. Note that because of the finite length of the buffer, multiple packets may be dropped over one time step when more packets arrive than the buffer can accomodate. Packets are dropped when the buffer contains v packets, and more than $B-v$ packets arrive during a time step. It can be shown that the average number of packet loss N_L is given by:

$$N_L = \sum_{v=0}^{B} \pi_v \Big(\sum_{n=0}^{min[v, N_m-(B-v)]} \Big(\sum_{k=B-(v-n)}^{N_m} (k - B + v - n) q_k \Big) z_{n,v} \Big), \tag{3.16}$$

where π_v is the steady-state probability for state v based on the new transition probability matrix in (3.2.1). The packet loss of the AMC-ARQ systems with the realistic Poisson traffic arrival pattern is shown in Fig. 3.5. The performance of SR is again the best, especially over the high SNR region. Further, it can be observed that the performance of the proposed system degrades over a wide range of average SNR. This is because the transmitter will have less chance to retransmit erroneous packets before they are dropped.

3.2.2.2 Average delay

The average queueing delay, defined as the number of time steps that a packet needs to wait before being transmitted, can be similarly determined by applying Little's law as in (3.8). The average length of the queue Q_a for realistic Poisson traffic can still be calculated using (3.9) but with π_v obtained from the transition probability matrix in (3.2.1). The average delay performance of the proposed AMC-ARQ system is presented in Fig. 3.6. As for the steady state traffic arrival pattern, the delay performance of the three strategies are very similar for the same value of λ. The average delay over the low SNR range decreases significantly from 5 to 2.5, when the traffic intensity (λ) is increased from 1 to 2. In the lower SNR range the buffer is always full due to the poor channel conditions, and most packets are dropped as a result. It should be noted that average packet delay here takes into account the delay of all packet in the queue including dropped packets. Hence the decrease in average delay with an increase in traffic intensity. On the average, packets will stay in the buffer for N_m/λ time steps. The variation in average delay with an increase in traffic intensity is less pronounced over the high SNR range.

3.2.2.3 Transmission efficiency

It can be seen that the physical layer spectral efficiency of the proposed transmission scheme for realistic Poisson traffic is the same as that for steady stream traffic given in (3.11). Noting that the average number of successfully received packets over one time slot is equal to $\lambda - N_L$, the average number of transmissions/retransmissions that the transmitter needs to perform per successfully received packet can be obtained as:

$$N_L = \frac{1}{\lambda - N_L} \sum_{i=1}^{M} P_i \sum_{v=1}^{B} \pi_v \min[i, v] \tag{3.17}$$

The transmission efficiency of the proposed AMC-ARQ system with a Poisson traffic arrival pattern is illustrated in Fig. 3.7 and 3.8. Fig. 3.7 plots the average number of packets transmitted during a timeslot with the three strategies for Poisson traffic. The spectral efficiency of the three ARQ strategies are very similar over all traffic intensities. The spectral efficiency of the three buffer management strategies improves and saturates to the maximum of λ packets per time slot. As such, the maximum spectral efficiency is limited by the traffic intensity, in addition to the fading channel condition.

The average number of packet transmissions per successfully received packet for the three strategies are plotted in Fig. 3.8. It can be seen that, SR exhibits the largest power efficiency among the three strategies. As for the steady stream case, the maximum number of transmissions per successfully received packet remains low for low average SNR ranges, peaks at a mid-SNR range and then decreases at higher values of average SNR. This is due to the lower number transmission attempts at low values of average SNR. At the higher SNR range, most of the packets will be delivered successfully upon the intitial transmission, and hence a lesser number of transmissions are required to successfully send a packet. It should also be noted that when λ increases from 1 to 2, the maximum number of packets transmissions per successfully received packet increases, and occurs at a higher average SNR value. When the traffic intensity increases, more packets will be dropped due to buffer overflow. Thus, the transmitter will have lesser opportunities to retransmit the erroneously received packets.

In this chapter, we analyzed the performance of the SNR-based AMC-ARQ transmission system. The results indicate that joint PR and AMC transmission provides good performance with low implementational complexity, as compared to SR and AR. It can also be seen that an increase in the buffer size leads to a significant performance improvement at the cost of increased average delay for packets in the queue, at lower SNR ranges.

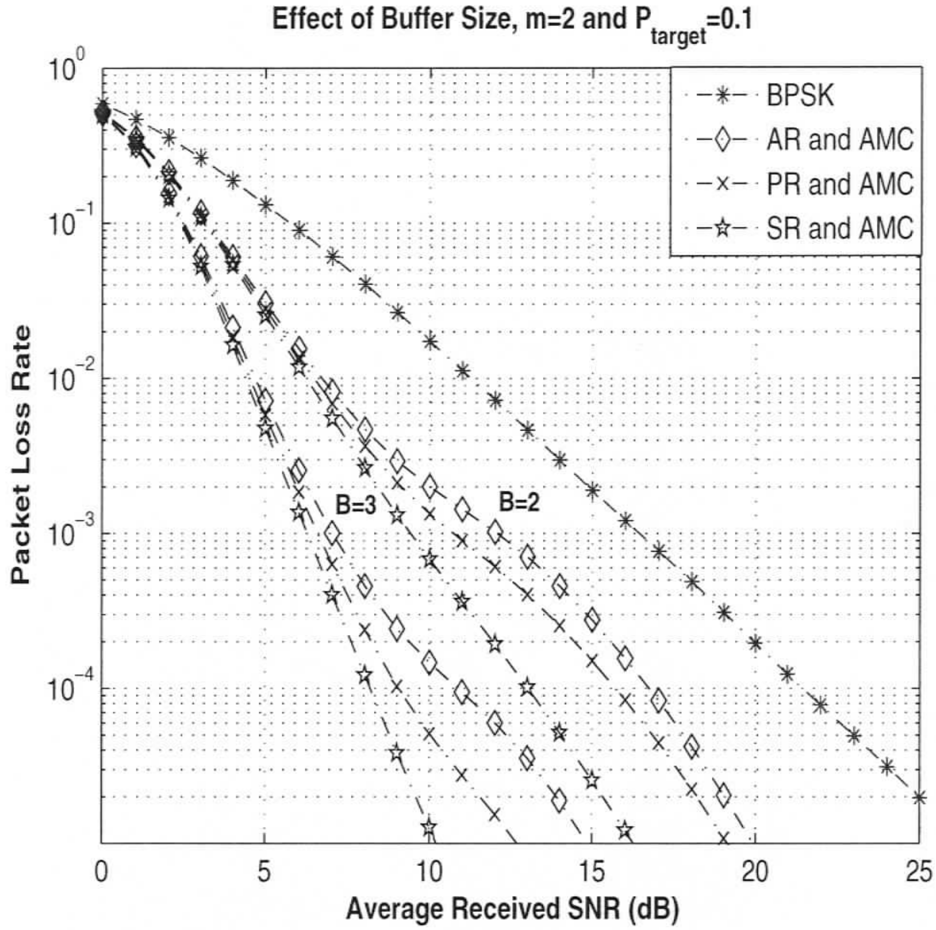

Figure 3.1. *Average packet error rate for steady stream traffic as a function of average path SNR for different buffer sizes.*

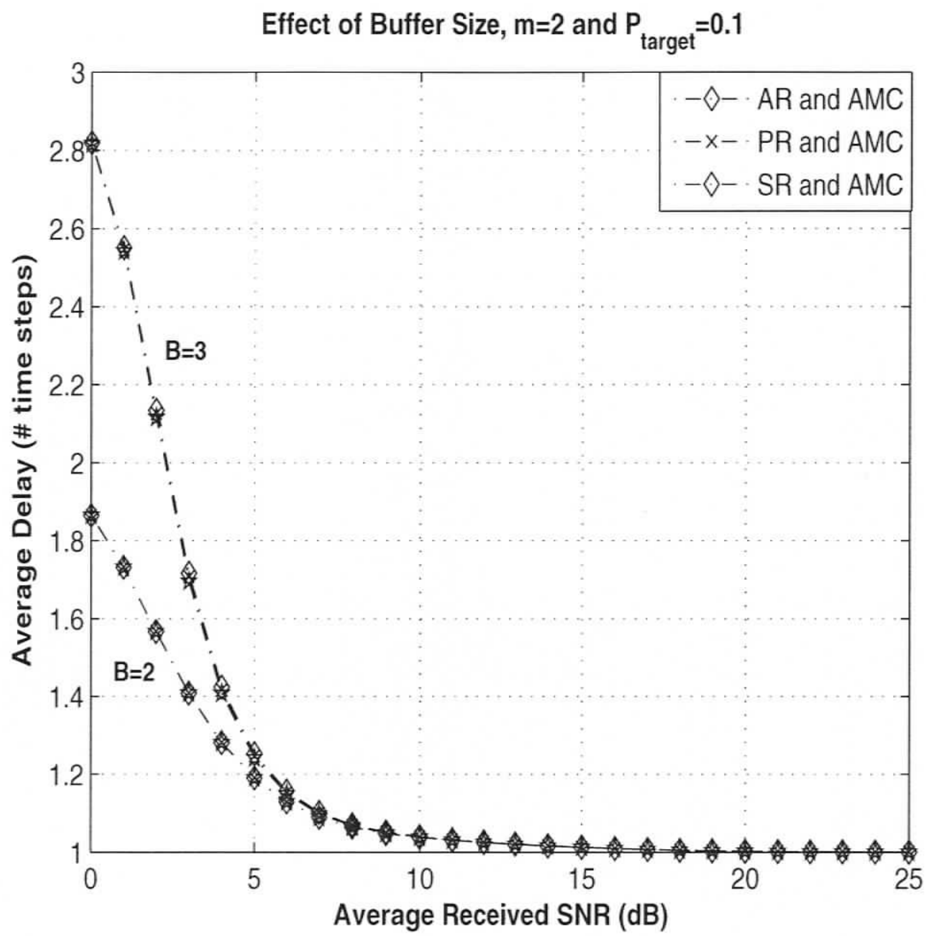

Figure 3.2. *Average wait-time for steady stream traffic as a function of average path SNR for different buffer sizes.*

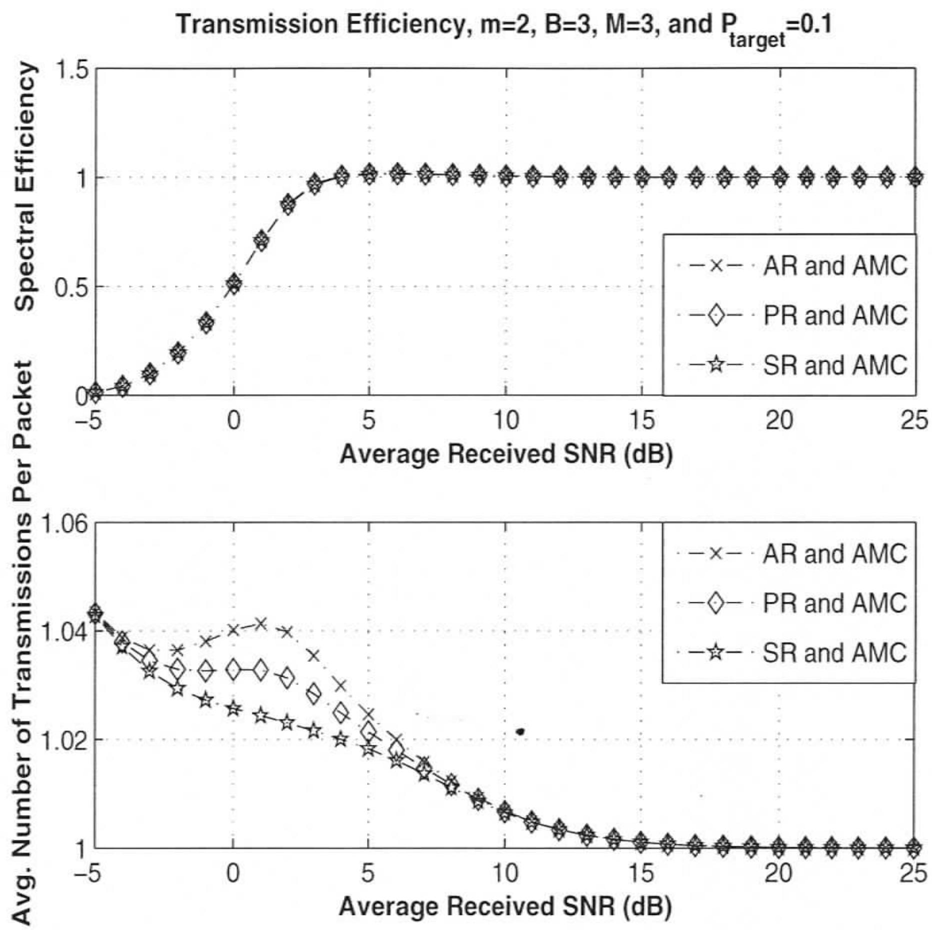

Figure 3.3. *Transmission efficiency for steady stream traffic.*

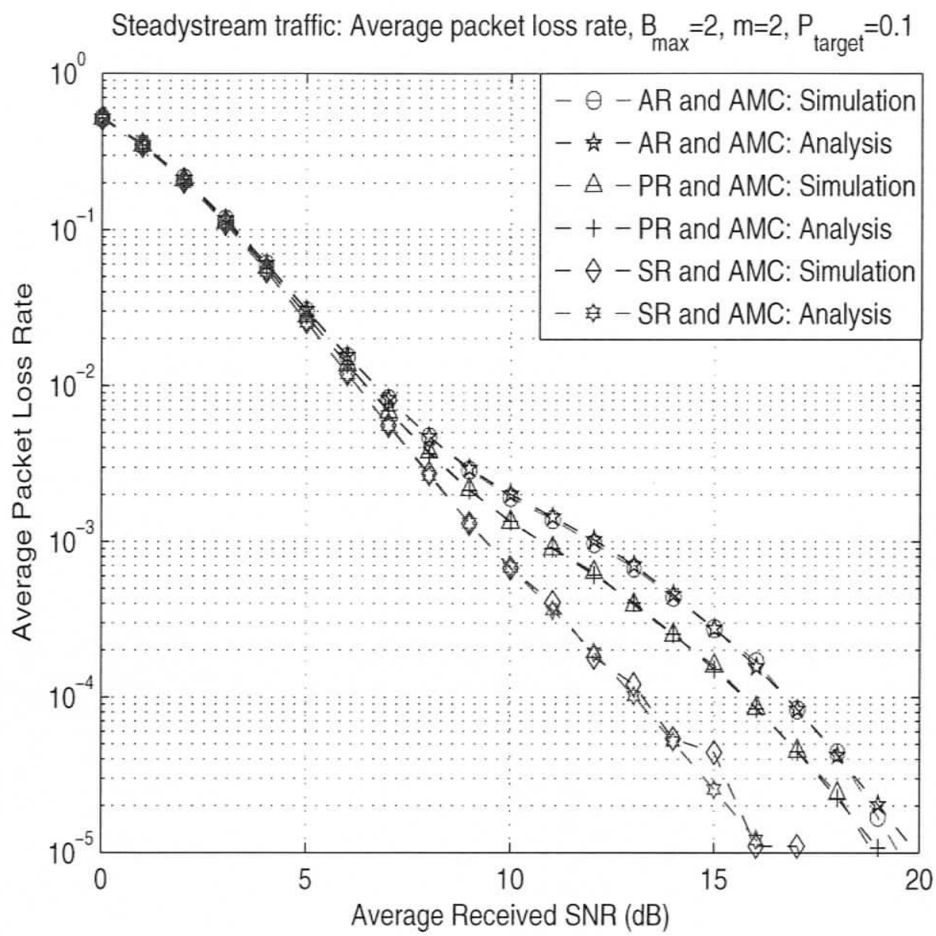

Figure 3.4. *Packet loss rate for steady stream traffic*

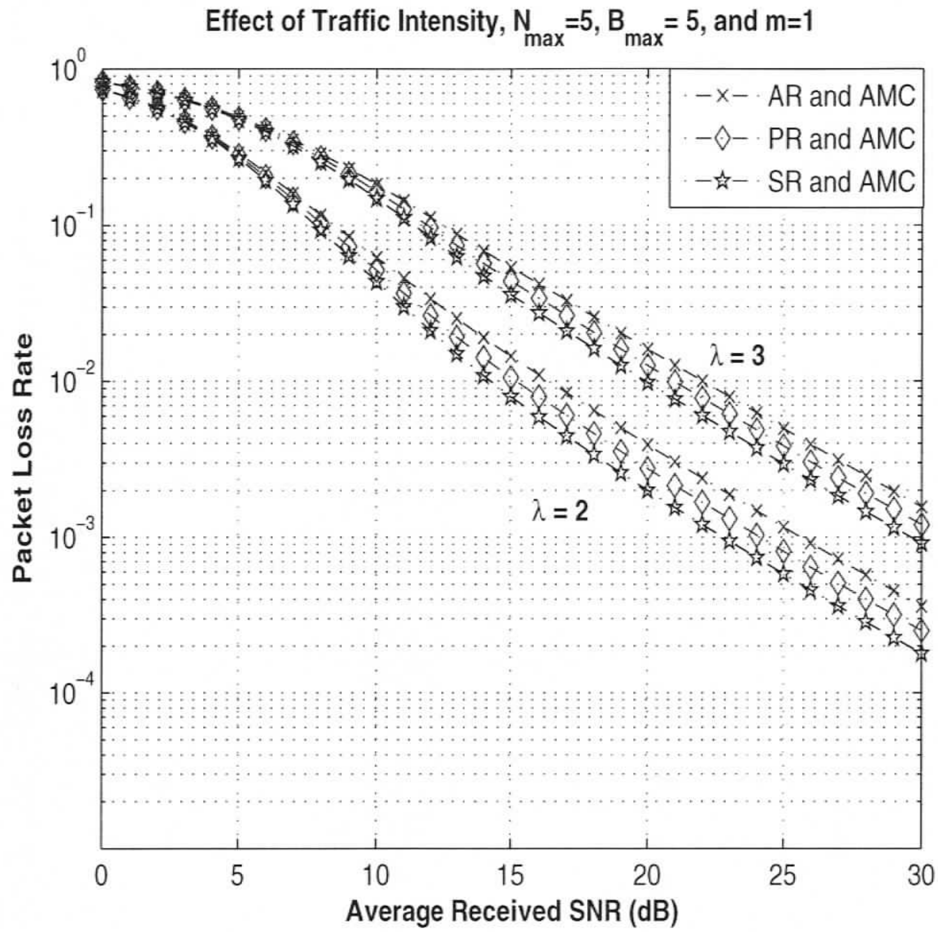

Figure 3.5. *Average packet error rate for realistic Poisson traffic as a function of average path SNR for different traffic intensities.*

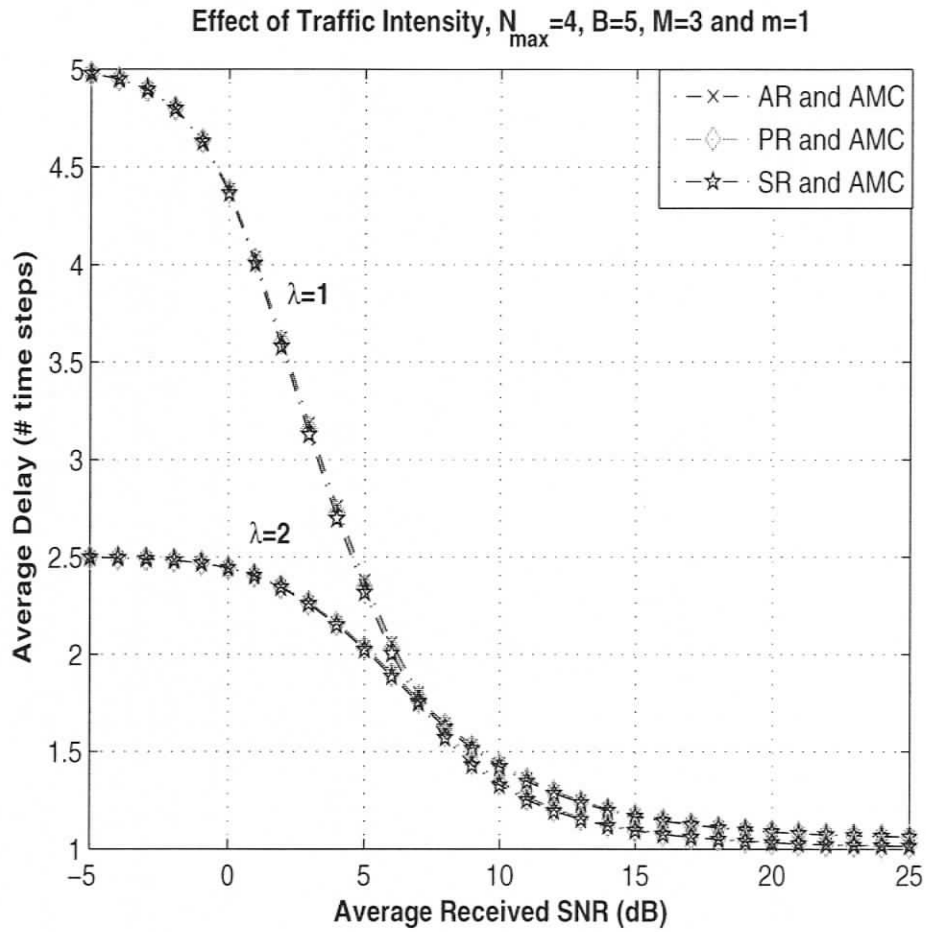

Figure 3.6. *Average packet delay for realistic Poisson traffic as a function of average path SNR for different traffic intensities.*

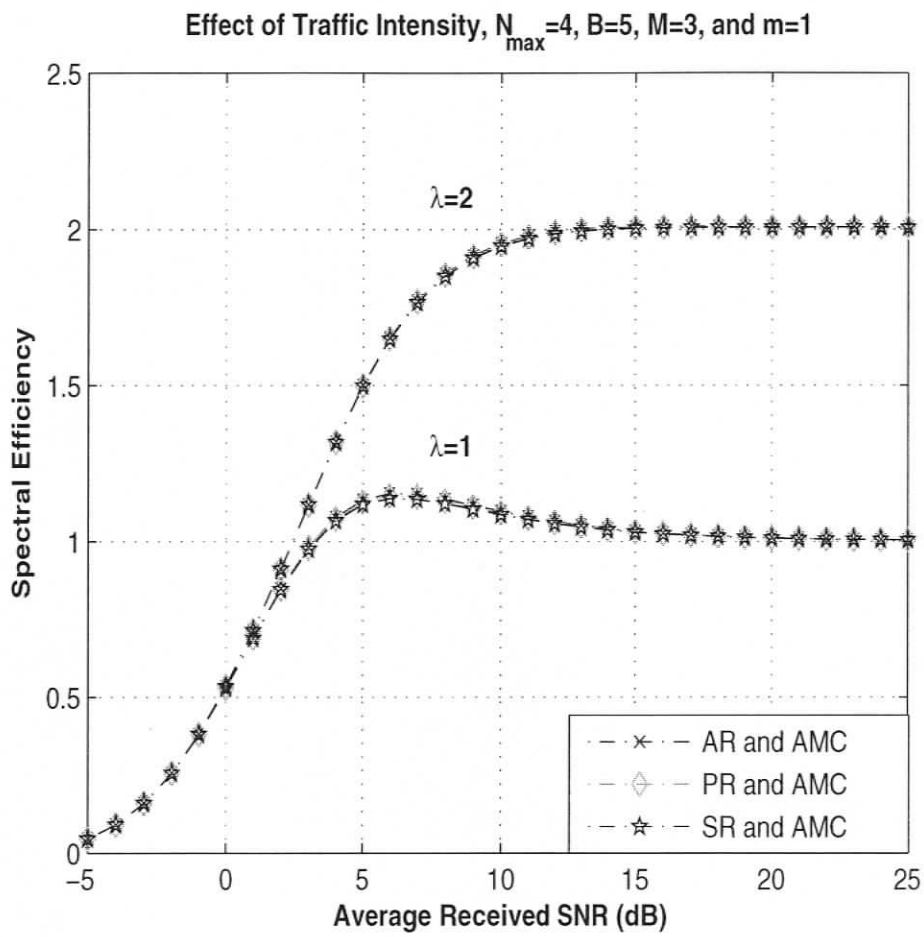

Figure 3.7. *Average spectral efficiency for realistic Poisson traffic as a function of average path SNR for different traffic intensities.*

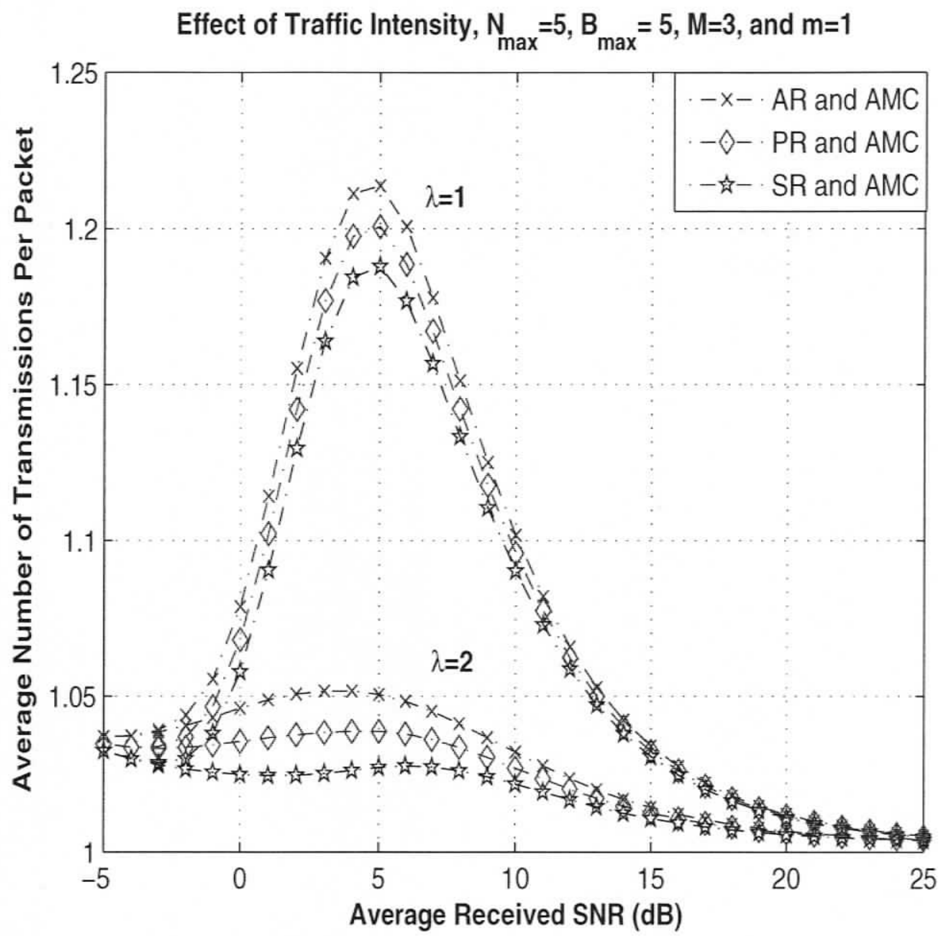

Figure 3.8. *Average number of transmissions per successful packet for realistic Poisson traffic as a function of average path SNR for different traffic intensities.*

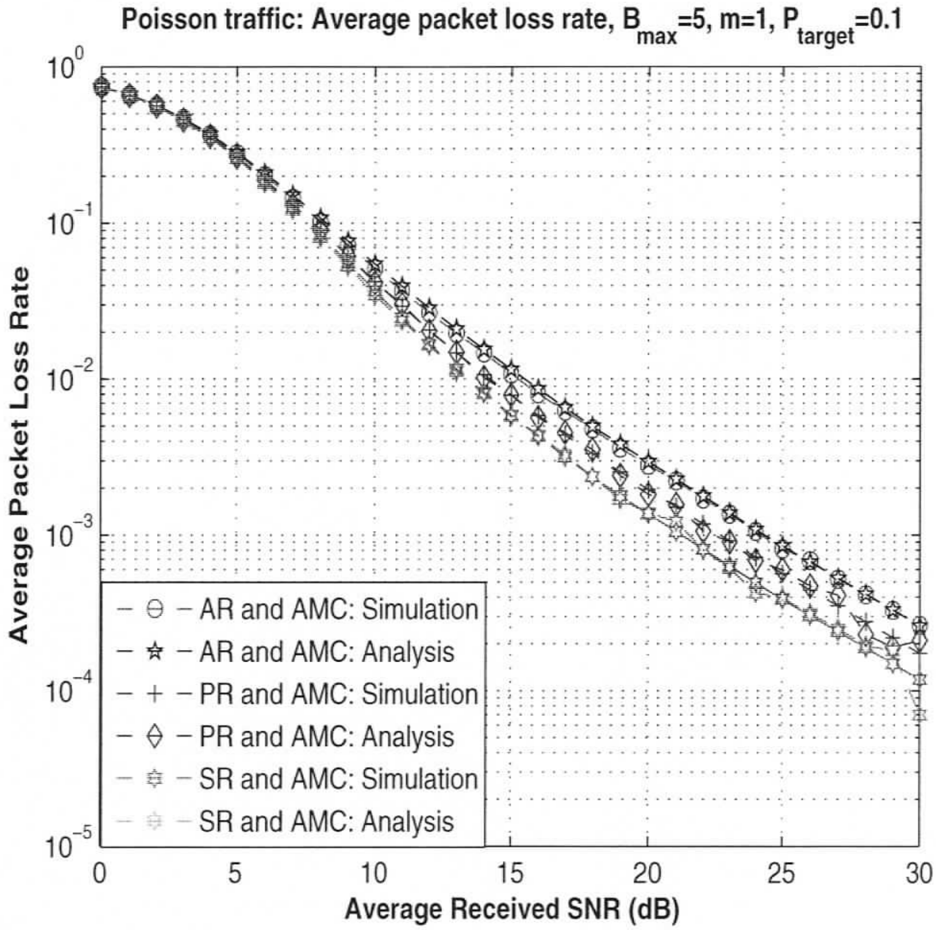

Figure 3.9. *Average packet loss rate for realistic Poisson traffic as a function of average path SNR for different traffic intensities.*

Chapter 4

Analysis of Buffer-assisted Joint AMC-ARQ Transmission

The SNR-based AMC technique that has been discussed so far selects the mode of transmission based solely on the fading channel condition. However, a channel-based mode selection strategy could lead to transmission at high data rates irrespective of the status of the buffer. This would result in a higher probability of error, thus reducing the reliability of transmissions. To address this drawback, we evaluate the performance of the buffer-assisted AMC-ARQ technique, where the mode of transmission is selected based on the fading channel conditions as well as the state of the finite-length buffer at the transmitter. The performance and efficiency of this new cross-layer strategy is evaluated in the following sections by making use of the Markov chain based approach developed in the previous chapter. We show that the buffer-assisted AMC-ARQ technique can effectively improve the performance and power efficiency of the low-complexity AR scheme while maintaining the same average wait time and spectral efficiency as the SNR-based AMC technique. It is also shown that a marginal improvement in performance can be acheived for the PR scheme. The use of the buffer-assisted mode selection strategy would improve the reliability and power efficiency of HSDPA at little cost in terms of protocol overhead.

4.1 Markov Chain-based Analysis

The Markov chain model defined in Chapter 3 can be used to model the dynamics of the transmitter buffer. Since the new mode selection strategy affects only the departure process, we will arrive at the same expressions for the transition probabilities as given in Eq. (3.14) and (3.15), while the probabilities $z_{n,v}$ for the three management schemes should be modified. In particular, it may happen that while the received SNR is in the region (γ_i, γ_{i+1}) the system will decide to use AMC Mode v, $v < i$ when there are just v packets currently in the buffer. It can be seen that the packet error rate that occurs when Mode v is selected is integrated between the SNR thresholds for Mode i. The average PER when using mode v for transmission while the received SNR is in the region (γ_i, γ_{i+1}) denoted by $\overline{PER}_{i,v}$, $i = 1, 2, \cdots, M$ and $v \leq i$, for Nakagami fading environment is given by:

$$\begin{aligned} \overline{PER}_{i,v} &= \frac{1}{P_i} \int_{\gamma_i}^{\gamma_{i+1}} per_v(\gamma) . \, p_\gamma(\gamma) d\gamma \\ &= \frac{1}{P_i} \frac{a_v}{\Gamma(m)} \Big(\frac{m}{\overline{\gamma}}\Big)^m \times \frac{\Gamma(m, b_v\gamma_i) - \Gamma(m, b_v\gamma_{i+1})}{(b_v)^m}, \end{aligned} \tag{4.1}$$

where $b_v = \frac{m}{\bar{\gamma}} + g_v$ and P_i is the probability that the received SNR is in the region (γ_i, γ_{i+1}), which was given in (2.5). It should be noted that this differs from the 'P_i' for SNR-based mode selection, where it represented the probability that Mode i is selected based on the instantaneous channel conditions. Note that when $i = v$, $\overline{PER}_{i,v}$ reduces to $\overline{PER}_i$. We now derive the analytical expression for $z_{n,v}$ for three buffer management schemes with the buffer-assisted mode selection scheme in the following.

4.1.1 AR scheme

In this case, either all $n > 0$ transmitted packets leave the queue or no packet leaves the queue. Note that when $0 < n < \min[v, M]$, the index of the selected AMC Mode, which is equal to n, is always smaller than the number of packets in the buffer. As such, the expression of $z_{n,v}$ for this case is the same as the one given in (3.2). For the cases of $n = v \leq M$ and $n = 0$, the system may select a lower AMC Mode than the fading channel condition allows due to the limited number of packets in the buffer. In particular, the AMC Mode v will always be used regardless of the SNR value for the case of $n = v \leq M$. Finally, after considering the $v < M$ case separately for $n = 0$, the expression of $z_{n,v}$ for AR scheme with the new mode selection strategy can be calculated as

$$z_{n,v} = \begin{cases} \sum_{i=v}^{M} P_i \left(1 - \overline{PER}_{i,v}\right)^v, & n = v \leq M; \\ P_n \left(1 - \overline{PER}_n\right)^n, & 0 < n < \min[v, M+1]; \\ P_0 + \sum_{i=1}^{\min[v,M]} P_i \left[1 - \left(1 - \overline{PER}_i\right)^i\right] & \\ \quad + \sum_{i=v+1}^{M} P_i \left[1 - \left(1 - \overline{PER}_{i,v}\right)^v\right], & n = 0; \end{cases} \tag{4.2}$$

where $\overline{PER}_{i,v}$ was given in (4.1).

4.1.2 PR scheme

With the PR scheme, n packets leave the queue if and only if the first n packets transmitted are correctly received and the $n + 1$th packet, if transmitted, is detected in error. Similar to the AR scheme, the expression of $z_{n,v}$ for the $n = v \leq M$ case is the same as the one given in (4.2). Note that when the number of packets v in the buffer is smaller than M, the Mode selection may be adjusted to just accommodate the number of packets in the buffer. We again consider the case that $v < M$ separately and arrive at the following expressions for the probability $z_{n,v}$ for PR scheme with the new mode selection strategy.

$$z_{n,v} = \begin{cases} \sum_{i=v}^{M} P_i \left(1 - \overline{PER}_{i,v}\right)^v, & n = v \leq M; \\ P_n \left(1 - \overline{PER}_n\right)^n & \\ \quad + \sum_{i=n+1}^{\min[v,M]} P_i \left(1 - \overline{PER}_i\right)^n \overline{PER}_i & \\ \quad + \sum_{i=v+1}^{M} P_i \left(1 - \overline{PER}_{i,v}\right)^n \overline{PER}_{i,v}, & 0 \leq n < \min[v, M+1]. \end{cases} \tag{4.3}$$

4.1.3 SR scheme

With SR scheme, only packets received in error during the current time slot are to be retransmitted. By considering the cases $v < M$ and $v \geq M$ separately, we can similarly obtain the expression of the probability $z_{n,v}$ for SR scheme with the new selection strategy as:

$$\begin{aligned} z_{n,v} &= \delta(n)\, P_0 + \sum_{i=n}^{\min[v,M]} P_i \binom{i}{n} (1 - \overline{PER}_i)^n (\overline{PER}_i)^{i-n} \\ &\quad + \sum_{i=v+1}^{M} P_i \binom{v}{n} (1 - \overline{PER}_{i,v})^n (\overline{PER}_{i,v})^{v-n}, \end{aligned} \tag{4.4}$$

where $\delta(\cdot)$ is the Delta function. With the transition probability matrix of the Markov chain thus determined, we can also study the performance of the buffer-assisted mode selection strategy based on the steady state behavior of the chain.

4.2 Numerical examples

After similar type of reasoning, we can arrive at the same analytical expressions for the packet loss rate, packet delay and transmission efficiency as given in the previous section, with the notion that the steady state probability vector be calculated using the new transition probability matrix determined above. In the following, we will examine the packet loss rate and power efficiency of this new mode selection strategy through numerical examples. In Fig. 4.1, we compare the packet loss rate of the transmission system with two different mode selection strategies for the three buffer management schemes under consideration. We set the buffer size B equal to 10 and Nakagami fading parameter m to 2. It can be seen that the buffer-assisted mode selection strategy leads to a smaller packet loss rate than the SNR-based strategy studied in previous section. We can also notice that the performance gain with the buffer-assisted strategy is most significant for the AR scheme. This indicates that the increased reliability obtained by using a lower AMC mode based on the buffer status is more beneficial to AR and PR schemes than for the SR scheme. On the other hand, it seems unnecessary to introduce the extra complexity of mode update, if the SR scheme is employed. The power efficiency of the buffer-assisted mode selection strategy is plotted in Fig. 4.2 in comparison with those of the SNR-based strategy. We can immediately observe that the buffer-assisted strategy significantly increases the power efficiency of the system by reducing the number of packet transmissions per successfully received packet, especially for the medium to high SNR range. We can also see that the power efficiency savings are most significant for the AR scheme, while a slight reduction in wasteful retransmissions is observed for the SR scheme. Fig. 4.3 and Fig. 4.4 compares the average delay and spectral efficiency for the two AMC strategies for a buffer size of 10. It can be seen that there is little variation in these two performance measures between the two AMC strategies.

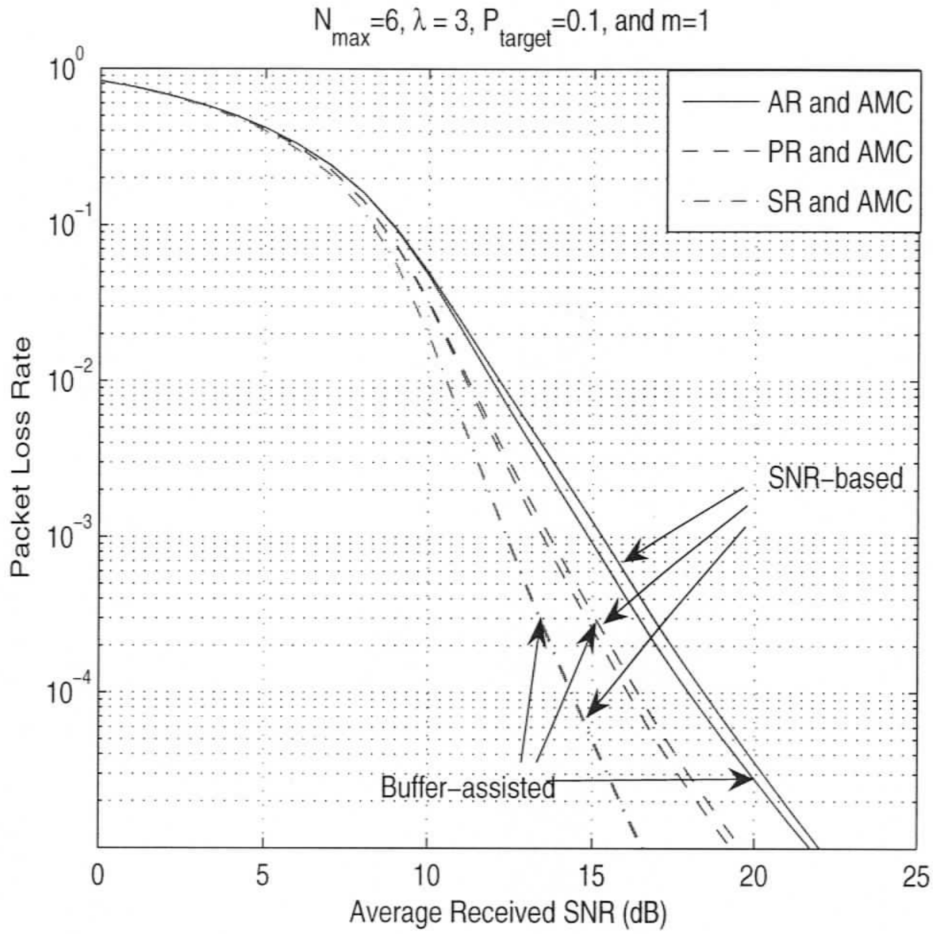

Figure 4.1. *Comparision of packet loss rate of SNR-based and buffer-assisted AMC strategies for three buffer management schemes (B=10).*

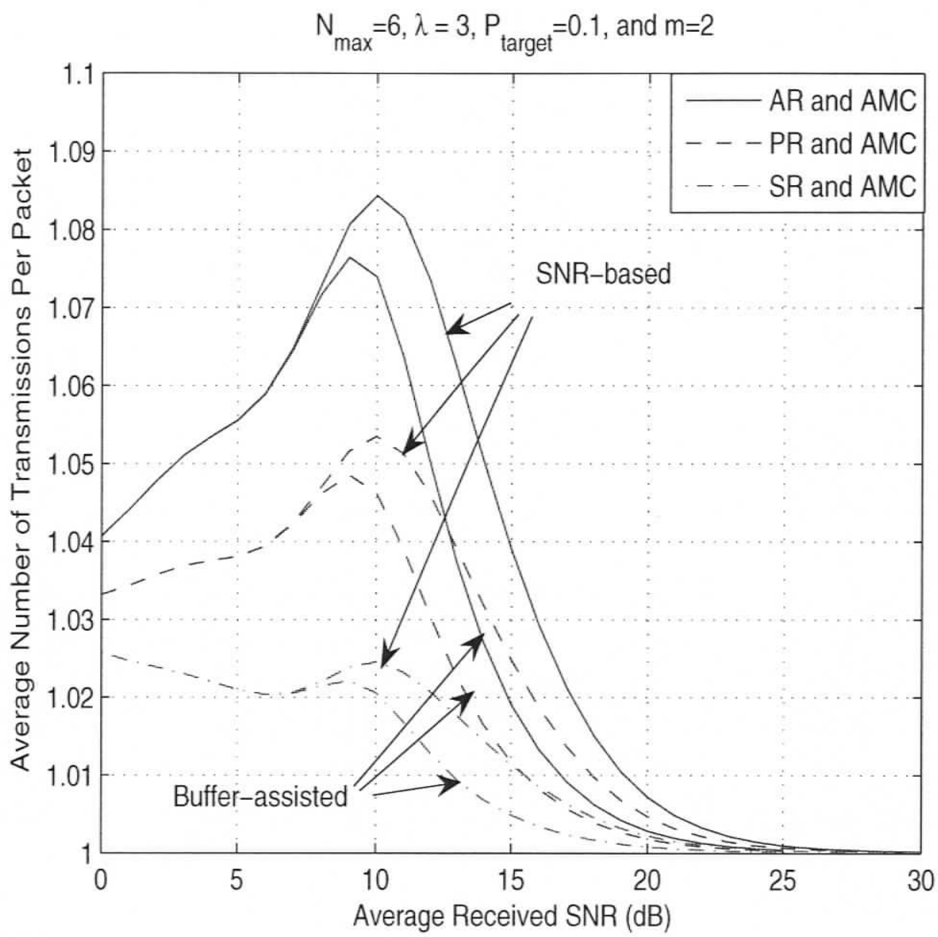

Figure 4.2. *Comparision of power efficiency of SNR-based and buffer-assisted AMC strategies for three buffer management schemes (B=10).*

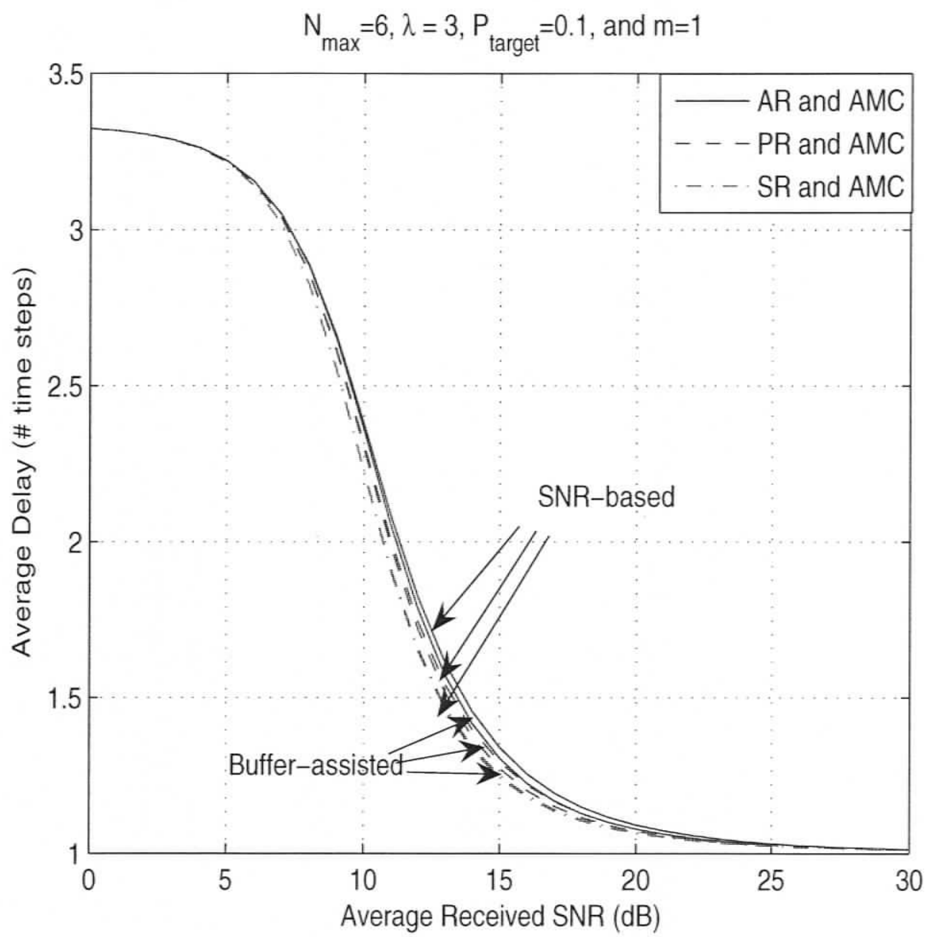

Figure 4.3. *Comparision of average wait time of SNR-based and buffer-assisted AMC strategies for three buffer management schemes (B=10).*

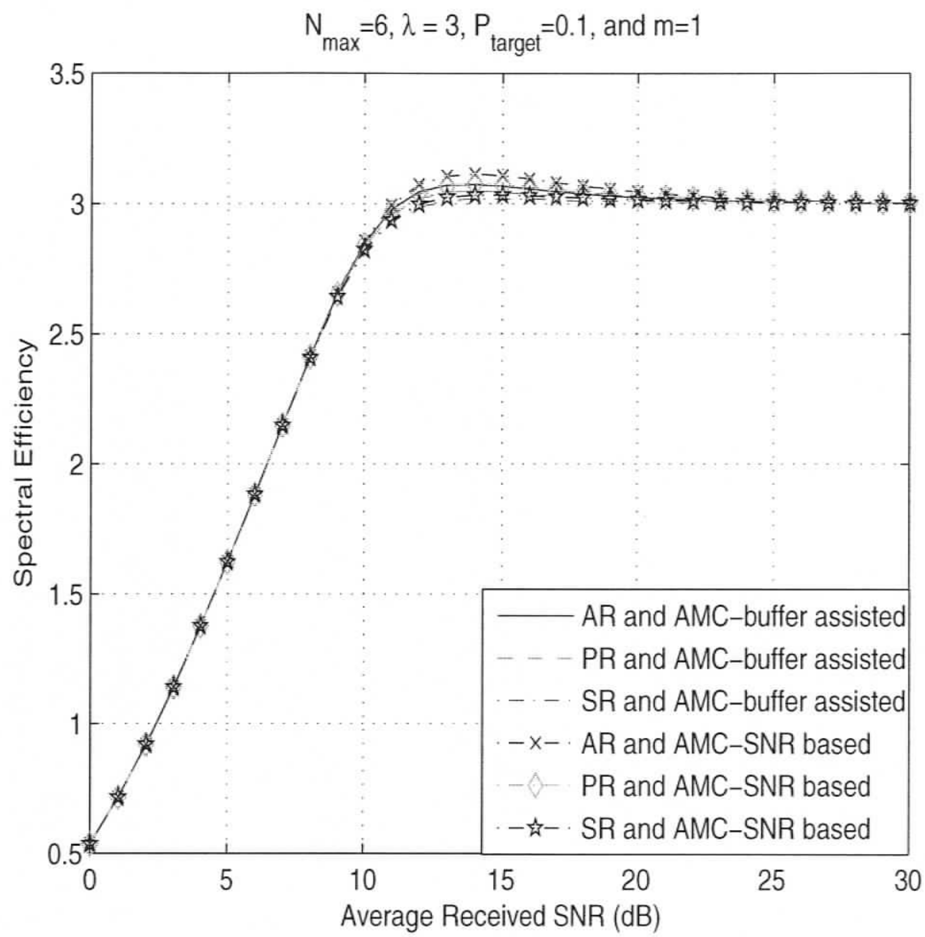

Figure 4.4. *Comparision of spectral efficiency of SNR-based and buffer-assisted AMC strategies for three buffer management schemes (B=10).*

Chapter 5

Concluding Remarks

5.1 Summary

The performance of AMC-ARQ wireless systems with finite-length buffer constraints at the transmitter have been analyzed using Markov chain theory. The packet loss rates, average wait time, power and spectral efficiency of AMC systems with different types of ARQ strategies have been analyzed and compared. The results indicate that PR is a good design choice for buffer management in AMC-ARQ wireless system, due to their improved performance with little increase in receiver complexity. The results also indicate that the average wait-time of packets in the queue increase with an increase in buffer size. The effects of increased average wait time for real time transmission, and algorithms to address these effects are possible areas of future work that are discussed in the following section.

In addition, a new buffer-assisted AMC technique was analyzed and the performance improvement was quantified and compared with that of an SNR-based AMC-ARQ system. The results indicate that the buffer-assisted AMC scheme increases the transmission reliability in AMC-ARQ wireless systems while maintaining the same performance as an SNR-based AMC strategy. This technique can be readily applied to the HSDPA standard for mobile communications.

5.2 Future Directions

Through this study, we identify the following possible future research directions.

5.2.1 QoS prioritized transmission

The performance analysis of AMC-ARQ wireless systems with finite length buffer constraints at the transmitter show that even a small increase in the buffer size can lead to an increase in average wait time for packets in the queue. The increased wait time at the buffer can be detrimental for the transmission of real-time traffic like video and VoIP, that have a finite 'time-to-live'(TTL) before which it should reach the destination. Packets that do not reach the destination within the TTL period are not useful. Therefore, we may consider the transmission of heterogeneous traffic through AMC-ARQ wireless systems with multiple quality of service prioritized buffers at the wireless transmitter to reduce the queueing delay for real-time packets.

We carried out a preliminary study of AMC-ARQ systems that have multiple priortized queues at the transmitter for different clasess of traffic. Specifically, the first queue services real-time traffic while a second queue deals with non real-time traffic. During a time-slot, packets from the first queue are transmitted with a higher priority. Fig. 5.1 shows the results of a preliminary simulation study of this system that compares the packet loss rate for the two queues for different ARQ techniques. It can be seen that the reduced wait-time for real-time traffic comes at the cost of higher packet loss rates for non-real time transmissions. Therefore, an optimum scheduling mechanism would ensure a low average wait time for real time packets, while simultaneously ensuring that the packet loss for non-real time traffic does not exceed a prescribed threshold.

This applies to the 3GPP HSDPA standard which makes use of AMC and a multi-channel stop and wait ARQ (SAW-ARQ) scheme. The current HSDPA specification does not provide for quality of service differentiated queueing at the wireless transmitter. Hence, a real-time packet arriving at the buffer of a wireless transmitter may suffer a significant amount of queueing delay before transmission over the wireless channel. The 3GPP Release 5 specifications that define HSDPA also introduces an IP based network architecture for mobile systems. One of the primary objectives of IP Multimedia Subsystem (IMS) is to facilitate voice over IP (VoIP) calls from mobile terminals. Inorder to acheive the said objective, it is pertinent that the queueing delay for real-time traffic be reduced for better voice quality.

5.2.2 Cross-layer Scheduling

Scheduling of users at a base station plays an important role in QoS provisioning. Conventionally, users are scheduled either using a round-robin technique that makes use of fixed slots for transmission, or by channel allocation based on the instantaneous SNR of each user. The second scheduling strategy leads to higher spectrum utilization. However, a scheduling technique that takes into account the QoS guaranteed for each user would be more effective for real-time transmissions. A new packet scheduling algorithm for HSDPA was proposed in [33] where users are scheduled based on the received channel condition, packet delays and different QoS requirements amongst different classes of traffic. A cross-layer design for multiuser scheduling at the data link layer, with each user employing AMC was analyzed in [16]. The cross-layer scheduler guaranteed prescribed QoS for admitted QoS-guaranteed users and achieved efficient bandwidth utilization. The use of a QoS optimized scheduling strategy together with AMC-ARQ and prioritized queueing would intutively lead to significant improvements in the ensuring guaranteed QoS for real-time transmission. The performance of this system could be analyzed using a similar Markov chain-based approach. This technique would be especially useful in HSDPA mobile communication systems.

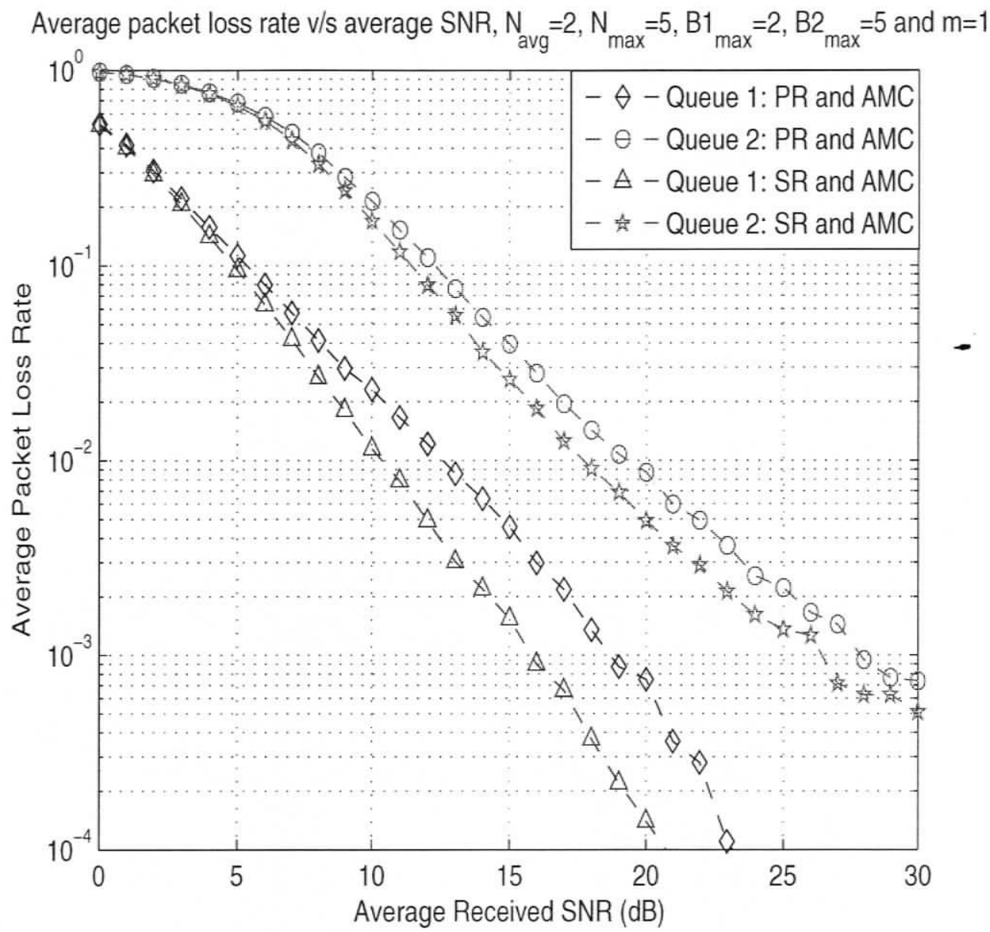

Figure 5.1. *Variation of packet loss in the two queues with buffer size*

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