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Link Adaptive Optimization Method based on Minimizing Packet Loss Rate in AOS Communication System

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Abstract

Aiming at the problem of packet loss of finite-length queues caused by the characteristics of data high burst, and finally leading to a decrease in throughput in the AOS space communication system, a link adaptive optimization method based on the minimization of the system packet loss rate is proposed. It combines the finite length queue, limited retransmission of the data link layer, and adaptive modulation coding of the physical layer. The system packet loss rate is used as an objective function. By solving the minimum system packet loss rate to reasonably assign the retransmission times and modulation encoding schemes, the system average throughput is improved finally. The theoretical analysis and simulation results show that this method can reduce the system packet loss rate by 80% and increase the system average throughput by 4.2% compared with the AMCA method. At the same time, it can reduce the system packet loss rate by 90% and increase the system average throughput by 11.1% compared with the AMCAFQS method.

Keywords: AOS; finite length queue; limited retransmission

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1. Introduction

With the rapid development of aerospace, manned spacecraft, and orbital space technology, human space activities have entered the era of space station. In addition to conventional telemetry, remote control, and tracking data, complex space tasks also put forward many difficult data processing requirements, because data transmission with complex data, high data rate, and high switching capacity needs dynamic management. The conventional Time Division Multiplexing (TDM) transmission mode and traditional measurement and control network cannot meet these requirements. Therefore, it is necessary to negotiate and formulate a new general standard to adapt to the development of space technology [1-4]. The Consultative Committee for Space Data System (CCSDS) was established in 1982 by the National Aeronautics and Space Administration (NASA), European Space Agency (ESA), and many space agencies in other countries [5]. CCSDS developed the system and standards of Advanced Orbiting System (AOS) to meet the needs of complex spacecraft [6].

AOS space communication has the characteristics of large data transmission, high burst, high delay, high error rate, and time-varying fading. These features put forward new requirements for the reliability and transmission efficiency of space communication [7-9]. In the study of the reliability and transmission efficiency of space communication, an adaptive modulation coding technology based on the throughput rate was proposed in [10]. It improved the link expected throughput rate. By selecting the different ARQ types and different MCS according to the different channel states, in [11], the HARQ protocol was simply modified, and the physical layer could be implemented without any additional signals, thus increasing throughput. A hybrid retransmission scheme based on code merging was proposed in [12]. It analyzed the influence of different modulation modes on the performance of the hybrid retransmission scheme. The throughput performance of the system was improved effectively by using the higher-order modulation scheme. A new hybrid automatic retransmission request transmission scheme was proposed in [13]. When the packet needed to be retransmitted in the scheme, it only transferred error segmented data by detection. The proposed new transmission scheme was obviously superior to the traditional scheme in the system throughput and average transmission times. The above documents studied in the reliability

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and transmission efficiency and improved the system performance. However, they did not take into account the characteristics of random arrival data flow and high data burst.

In view of the characteristics of random arrival data flow and high data burst, the joint effect of finite length queue and AMC was analyzed and a Finite State Markov Chain (FSMC) was established in [14]. The packet loss rate and average throughput were obtained by the FSMC model. Minimizing the system packet loss rate as a utility function, an AMC algorithm was proposed to improve the average throughput of the system. In [15], the impact of ARQ technology on the combination of physical layer AMC and data link layer queues was studied, and an AMC Algorithm for Federated Queue Status (AMCAFQS), which brings an increase in average throughput, was proposed. The above documents only considered the characteristics of high data burst and introduced queue to solve the problem. They brought benefits to the system performance, but infinite retransmission caused large delays. Therefore, this paper proposes a Link Adaptive Optimization (LAO) based on the minimization of the system packet loss rate. Limited retransmission (LR) is combined with finite length queue (FLQ) and adaptive modulation coding (AMC). The system packet loss rate is used as the objective function, and the retransmission times are rationally allocated by solving the minimum system packet loss rate. Simulation results show that LAO can reduce the system packet loss rate and improve the system average throughput.

2. Joint Optimization of LR, FLQ, and AMC

Considering the efficiency and reliability of data transmission in the AOS space communication system, a joint optimization diagram of AOS transmission system is presented, as shown in Figure 1.

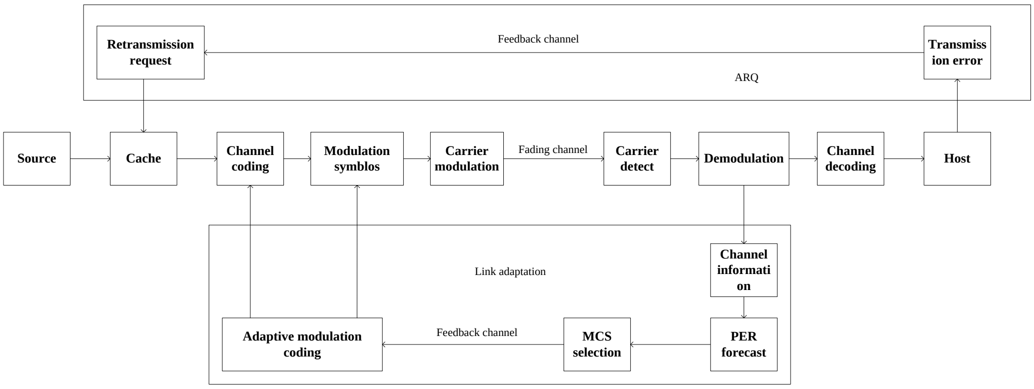

Figure 1. Optimization diagram of LR, FLQ, and AMC in AOS system

In Figure 1, the transmitter uses a finite length queue by first in, first out (FIFO). The source data enters a finite length queue, and then it goes through channel coding, symbol modulation, and carrier modulation. Finally, it is transmitted in the fading channel. At the receiver, the received signal goes though carrier wave detecting, demodulating, and decoding. In the adaptive modulation coding, the receiver adaptively adjusts the Modulation and Coding Scheme (MCS) based on the real-time channel state information. MCS selection is transmitted back to the transmitter through the feedback channel. AMC selection is achieved after Packet Error Rate (PER) prediction. In the automatic retransmission requests, the wrong data will be marked, and requested retransmission goes through the feedback channel to the transmitter. The packets in the cache of the transmitter will not be removed immediately, until the transmitter receives the correct feedback information. If the receiver is successfully decoded, the correct feedback information will be sent to the buffer of the transmitter. The transmitted packets will be deleted from the cache immediately.

3. Establishment of Objection Function

In the AOS space communication system, the system overall throughput performance is affected by the packet loss factor. The packet loss rate is affected by two factors: queuing packet loss and transmission error packet loss. a) Queuing packet loss: Packets are put in the cache queue at the transmitter before sending. If the queue is full, packets will be discarded, resulting in queuing packet loss. b) Transmission error packet loss: Due to the adverse effects of channel fading or interference, the receiver cannot correctly decode the received packets. After the maximum retransmission times, the packets are still decoded wrong. The packets will be discarded. Therefore, it is necessary to consider the effect of queuing packet loss and transmission error packet loss, reduce the system comprehensive packet loss rate, improve the throughput, and optimize the performance of the AOS communication system.

3.1. Objective Function

When the packet is not dropped by the queue and is received correctly through the fading channel, it indicates that the packet is received correctly from source to host. Let P_d stand for queuing packet loss rate. A packet from the source is correctly received by the subscriber, only if it is not dropped from the queue (with probability $1-P_d$), and it is correctly received through the fading channel (with probability $1-P$). Hence, the probability that packets can be received correctly is $(1-P_d)(1-P)$. The expression of system packet loss rate ξ is Equation (1).

$$\xi = 1-(1-P_d)(1-P) \tag{1}$$

Therefore, the system packet loss rate is taken as the objective function. At the same time, the number of retransmission is considered, and the problem is translated into the minimum value of the objective function.

Let λT_f denote the packet arrival rate upon the queue at the base station. The expression of the system average throughput is Equation (2).

$$\eta = \left(\lambda T_f\right)\left(1-\xi\right) = \left(\lambda T_f\right)(1-P_d)(1-P) \tag{2}$$

We know from Equations (1) and (2) that the key is to find P_d and P to evaluate the system performance in terms of packet loss rate and average throughput. The transmission error rate and queuing packet loss rate will be calculated below.

3.2. Calculation of Transmission Error Packets

Due to the adverse effects of channel fading or interference, the receiver cannot correctly decode and the packets will be discarded. Adaptive modulation coding and automatic retransmission request are adopted to reduce the number of error packets and transmission error rate.

3.2.1. AMC

MCS is a combination of modulation and coding. Five schemes of modulation and coding are adopted in this paper. In order to maximize the data rate and guarantee the specified packet error rate P_0, AMC selects the appropriate modulation and coding mode according to the channel change, so as to maximize the data rate.

It assumes the transmission power is constant and partitioned. The entire SNR range will be divided into $N+1$ non-overlapping consecutive intervals. N denotes the total number of available modulation coding schemes. Let $\{\gamma_n\}_{n=0}^{N+1}$ denote the boundary point of the subinterval and $\gamma \in [\gamma_n, \gamma_{n+1})$ denote the selection mode n. To avoid deep channel fades, no data are sent when $\gamma \in [\gamma_0, \gamma_1)$, which means $n=0$ and $R_0 = 0$. The selection of MCS requires determining the boundary points of the subinterval. The specific steps are as follows.

Step 1 Set $n = N,$ and $\gamma_{N+1} = +\infty$.

Step 2 For each *n*, search for the unique $\gamma_n \in [\gamma_0, \gamma_{n+1})$ and $\overline{PER_n} = P_0$.

Step 3 If $n > 1$, set $n = n-1$, and go to Step 2; otherwise, go to Step 4.

Step 4 Set $\gamma_0 = 0$.

Therefore, we get the SNR region $[\gamma_n, \gamma_{n+1})$. The SNR region $[\gamma_n, \gamma_{n+1})$ corresponding to transmission mode n constitutes the channel state indexed by n.

3.2.2. Joint AMC and ARQ

Considering the packet error, the MCS scheme used in retransmission is uncertain. Therefore, the average packet loss rate is calculated before we add the effect of retransmission on the packet loss rate.

SNR γ is a random variable whose probability density function is expressed in Equation (3).

$$p_\gamma(\gamma) = \frac{m^m \gamma^{m-1}}{\bar{\gamma}^m \ \tau(m)} \exp(-\frac{m\gamma}{\bar{\gamma}}) \tag{3}$$

Parameter m is the channel fading parameter $(m \geq \frac{1}{2})$.

According to the MCS of AMC, the probability of $MCSn$ being selected is shown in Equation (4).

$$\Pr(n) = \int_{\gamma_n}^{\gamma_{n+1}} p_\gamma(\gamma)\mathrm{d}\gamma = \frac{\tau(m, \frac{m\gamma_n}{\bar{\gamma}}) - \tau(m, \frac{m\gamma_{n+1}}{\bar{\gamma}})}{\tau(m)} \tag{4}$$

$\tau(m)$ is a gamma function and $\tau(m,x) = \int_x^\infty t^{m-1}\exp(-t)\mathrm{d}t$ is the complementary incomplete gamma function. γ_n and γ_{n+1} are the upper and lower boundary points corresponding to $MCSn$ respectively.

In the AMC transmission scheme, the average transmission error rate $\overline{PER_n}$ in $MCSn$ meets the conditional probability model. The calculation is shown in Equations (5) and (6).

$$\begin{aligned}\overline{PER_n} &= \frac{1}{\Pr(n)} \int_{\gamma_n}^{\gamma_{n+1}} a_n \exp(-g_n\gamma) p_\gamma(\gamma)\mathrm{d}\gamma \\ &= \frac{1}{\Pr(n)} \frac{a_n}{\tau(m)} (\frac{m}{\bar{\gamma}})^m \frac{\tau(m, b_n\gamma_n) - \tau(m, b_n\gamma_{n+1})}{(b_n)^m}, \quad n = 1, \cdots, N\end{aligned} \tag{5}$$

$$b_n = m / \bar{\gamma} + g_n \tag{6}$$

The average transmission error rate $\overline{PER}$ is shown in Equation (7).

$$\overline{PER} = \frac{\sum_{n=1}^{N} R_n \Pr(n) \overline{PER_n}}{\sum_{n=1}^{N} R_n \Pr(n)} \tag{7}$$

We consider the effect of the transmission error rate on the retransmission mechanism. Assume that the value of the maximum retransmission number $N_{\max}$ is from 0 to 4. The average error rate of the single transmission is $\overline{PER}$. Because the success rate of the packet transmission in the channel is mutually independent, the transmission error rate P that combines AMC and LR can be expressed as Equation (8).

$$P = 1 - \sum_{i=1}^{N_{\max}+1} \overline{PER}^{i-1} (1 - \overline{PER}) \tag{8}$$

3.3. Calculation of Queue Packet Loss

At the beginning of t unit time, there are U_{t-1} packets in the queue. At this time of the channel state, the transmitter can send C_t packets. When the time t ends, the number of packets arriving in the period is A_t. The C_t packets sent from the queue will not be deleted from the queue immediately but instead wait for the feedback information from the receiver. After adding the incoming A_t packets, there are U_t packets in the queue. We will analyze the arrival process, queue service process, and queue state recursion of packets respectively and then construct a finite state Markov chain. The queuing packet loss is related to the selection of queue length, retransmission number, and modulation coding. It is necessary to consider the effect of finite length queue, finite retransmission, and adaptive modulation coding on the queuing packet loss.

3.3.1. Arrival Process

The arrival process A_t is independent of queue status and channel state, which is expressed in Equation (9). We assume that A_t is Poisson distributed with parameter λT_f. The expectation of A_t satisfies $E\{A_t\} = \lambda T_f$. Let T_f denote unit time and $\Pr(A_t = a)$ denote the probability of the number of unit time packets arriving, which is in document [14].

$$\Pr(A_t = a) = \begin{cases} \dfrac{\left(\lambda T_f\right)^a \exp(-\lambda T_f)}{a!}, & \text{if } a \geq 0 \\ 0, & \text{otherwise} \end{cases} \tag{9}$$

3.3.2. Queue Service Process

The AMC module provides a dynamic service process for the queue. The number of packets per unit time is different. For $MCSn$, the number of packets transmits within the unit time is expressed in Equation (10), and b denotes the resource allocation parameters.

$$C_n = bR_n \tag{10}$$

Without considering the impact of ARQ, the queue service process is analyzed. The AMC module dynamically provides $N+1$ service status $\{C_n\}_{n=0}^{N}$ for the queue. When the channel is in the state n, we select $MCSn$, corresponding to the service state C_n. The service state changes with the change of channel state, and the evolution of service state can be represented by service process C_t. Here, we use the state transition matrix of FSMC in document [14] to model this service process.

Considering the influence of finite ARQ on queues, retransmission error packets are stuck in the cache queue. Therefore, retransmission times have a greater impact on queues. For simplicity, we assume that the transmission capacity of the queue is bR_n without considering retransmission. If we consider the influence of retransmission times, the average transmission times will be calculated, and the transmission capacity of the queue is shown in Equations (11) and (12).

$$C_n = \frac{bR_n}{\overline{N}} \tag{11}$$

$$\overline{N} = (1 + \overline{PER} + \overline{PER}^2 + \cdots + \overline{PER}^{N_{\max}}) = \frac{1 - \overline{PER}^{N_{\max}+1}}{1 - \overline{PER}} \tag{12}$$

3.3.3. Queue State Recursion

The queue state recursion model is shown in Figure 2.

Assuming the size of the queue cache is K, it can accommodate up to K packets at most. With the influence of ARQ, the recursion of queue state is shown in Equation (13).

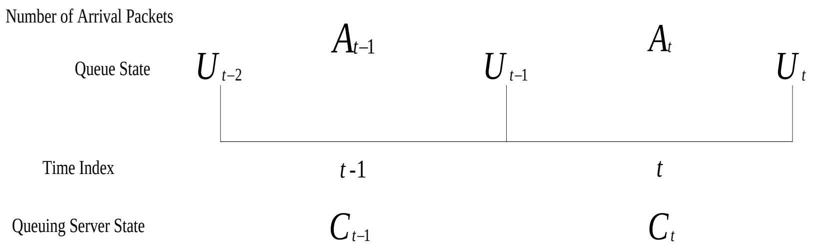

Figure 2. Queue state recursion model

$$U_t = \min\left\{Q, \max\left\{U_{t\text{-}1} - C_t\right\} + A_t\right\} \tag{13}$$

Equation (12) can see that the queue state depends on $\left(U_{t-1}, C_t\right)$. We create a FSMC process using a state pair $\left(U_{t-1}, C_t\right)$ that includes queue status and service status. Then, the FSMC process is analyzed and the joint stationary distribution of $\left(U_{t-1}, C_t\right)$ is obtained.

The steady-state distribution of $P(U = u, C = c)$ can be calculated from Equation (14).

$$\pi = \pi P, \quad \sum_{u \in U, c \in C} \pi_{(u,c)} = 1 \tag{14}$$

3.3.4. Calculation Procedure of Queuing Packet Loss Rate

Let D_t denote the number of packets discarded in time t, and D_t is related to U_{t-1}, C_t, and A_t. The expression is Equations (15) and (16).

$$D_t = \max\left\{0, A_t - K + \max\{0, U_{t-1} - C_t\}\right\} \tag{15}$$

$$\begin{aligned} E\left\{D\right\} &= \sum_{a \in A, u \in U, c \in C} DP(A = a, U = u, C = c) \\ &= \sum_{a \in A, u \in U, c \in C} \left[\max\left\{0, a - K + \max\left\{0, u - c\right\}\right\} P(A = a)\, P(U = u, C = c)\right] \end{aligned} \tag{16}$$

Based on Equations (11) to (16), the queuing packet loss rate is obtained in Equation (17).

$$P_d = \lim_{T \to \infty} \frac{\sum_{t=1}^{T} D_t}{\sum_{t=1}^{T} A_t} = \frac{E\left\{D\right\}}{E\left\{A\right\}} = \frac{E\left\{D\right\}}{\lambda T_f} \tag{17}$$

4. Link Adaptive Optimization Method based on Minimizing Packet Loss Rate

It is assumed that the Doppler frequency shift f_d, the average received SNR $\bar{\gamma}$, the channel fading parameter m, the fixed resource allocation parameter b, the receiving end queue length K, and the unit time packet arrival rate λT_f are known. According to Formula (2), the throughput maximization is equivalent to the system packet loss rate minimization. From Formula (1), we can see that the packet loss rate includes two parts: transmission error rate and queuing packet loss rate. From Formula (8), we can see that the transmission error rate will be reduced with the increase in the maximum number of retransmission. From the form of (11) to (17), it can be seen that the queuing packet loss rate is affected by the maximum retransmission number. The queuing packet loss rate will increase with the increase in the maximum retransmission number. The maximum retransmission times will affect the transmission error rate and queuing packet loss rate. In the range of maximum retransmission times, the number of retransmission is adjusted to balance the influence of the transmission error rate and queuing packet loss rate and minimize the system packet loss rate. The retransmission number distribution is shown in Figure 3.

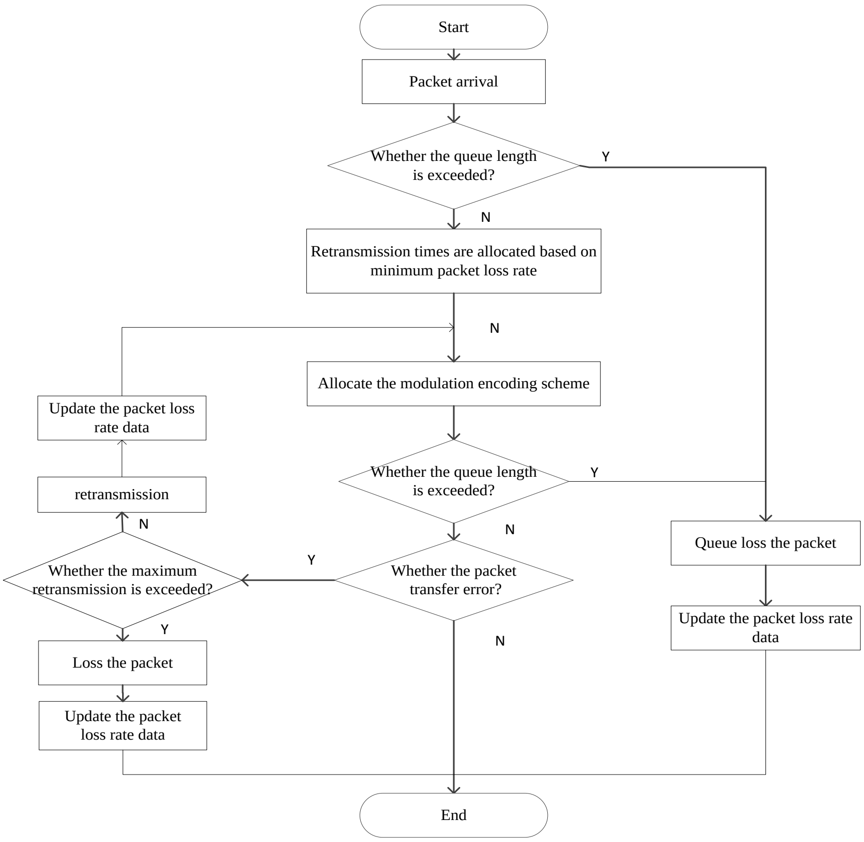

Figure 3. Retransmission number distribution flowchart

The specific steps for the selection of retransmission times are as follows.

Step 1 When the packet arrives, it determines whether the data is more than a finite length queue.

Step 2 If the packet exceeds the queue length, the queue loses the packet to update the packet loss data and jumps to step 10; otherwise, skip to step 3.

Step 3 Distribute retransmission times based on the minimum system packet loss rate.

Step 4 Transmit data based on the channel information distribution modulation coding scheme.

Step 5 Judge whether the data is more than a finite length queue again.

Step 6 If the packet exceeds the queue length, the queue loses the packet to update the packet loss data and jumps to step 10; otherwise, skip to step 7.

Step 7 Determine whether data transmission is wrong. If there is no error, jump to step 10; otherwise, skip to step 8.

Step 8 Determine whether the number of retransmission exceeds the maximum number of retransmission. If it exceeds the maximum number of retransmission, the queue loses the packet to update the packet loss data and jumps to step 10; otherwise, skip to step 9.

Step 9 Retransmit, update packet loss rate data, and then jump to step 4.

Step 10 End.

5. Experimental Simulation

5.1. Setting of Simulation Parameters

The above contents are simulated, and the simulation parameters are shown in Table 1.

Table 1. Simulation parameters

Simulation parameters	Parameter values
Doppler frequency shift f_d	10HZ
Channel fading parameter m	1
Modulation coding format	QPSK1/2, QPSK3/4,16QAM9/16,16QAM3/4,64QAM3/4
Queue length K	20 data packets
Poisson arrival rate λT_f	2 packet/ unit time
Packet length N_p	1080 bit
Resource allocation parameter b	2 data packets

5.2. Simulation Results and Analysis

The numerical results in Figure 4 show that system packet loss first decreases, then increases, and finally becomes stable as retransmission times increase. Because the decrease in the transmission error rate is greater than the increase in the queuing packet loss rate, the system packet loss rate decreases in the front part of these curves. The system packet loss rate increases in the middle part of these curves because the transmission error rate becomes very small as retransmission times increase. Meanwhile, the queuing packet loss rate becomes larger as retransmission times increase and the leading factor that affects the system packet loss rate is the queuing packet loss rate. In the latter part of these curves, the system packet loss rate finally tends to be stable, because the transmission error rate is close to zero. Then, the probability of retransmission is also near zero, and the queuing packet loss rate tends to be stable.

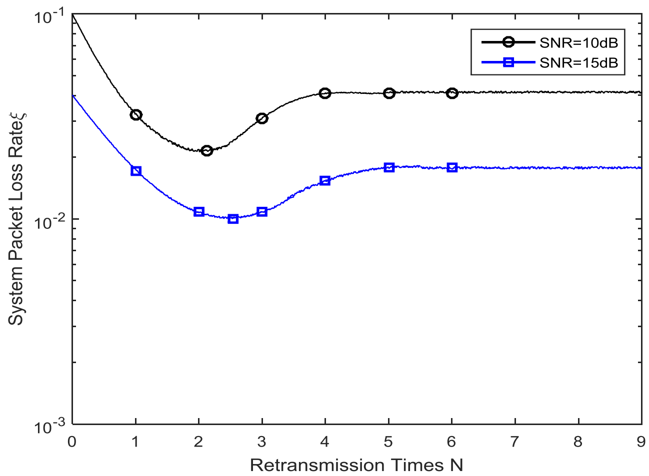

Figure 4. System packet loss rate

As shown in Figure 5, the LAO method is compared with the AMCA and the AMCAFQS method in the system packet loss rate. It can be seen from the diagram that the system loss rate of the three methods decreases with the increase of the signal to noise ratio. The LAO method drops faster than the AMCA method because the LAO method introduces limited retransmission, which can effectively reduce the transmission error rate and the system packet loss rate. The LAO method drops faster than AMCAFQS because AMCAFQS introduces infinite retransmission. Although the infinite retransmission can effectively reduce the transmission error rate, it also causes an increase in the queuing packet loss rate of the finite length queue. The three approaches are close to the same stability value when the signal-to-noise ratio is good, and the transmission error rate is close to zero. Then, the probability of triggering retransmission is also close to zero, and the queuing packet loss rate does not need to consider the effect of the queue loss caused by retransmission.

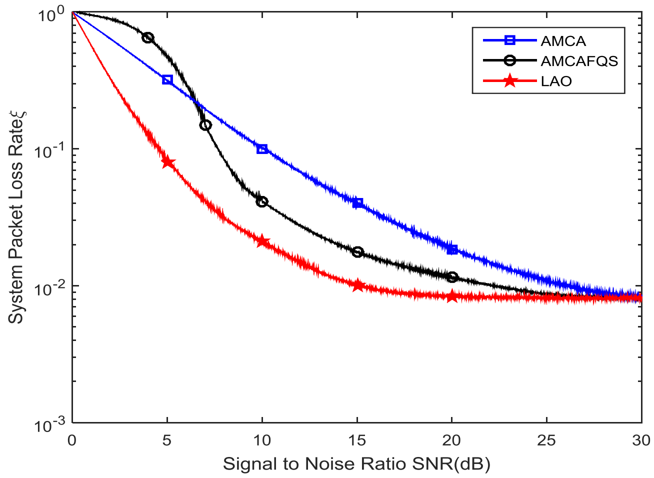

Figure 5. System packet loss rate versus signal-to-noise ratio

As shown in Figure 6, the LAO method is compared with the AMCA and AMCAFQS methods in the system average throughput. From the diagram, it can be seen that the system average throughputs of the three methods are all increasing with the increase in the signal to noise ratio. The LAO method rises faster than the AMCA method because the LAO method introduces the finite retransmission. The finite retransmission can effectively reduce the system packet loss rate, and the system average throughput is effectively increased. The LAO method rises faster than the AMCAFQS method because the AMCAFQS method introduces the infinite retransmission. Although the infinite retransmission can effectively reduce the transmission error rate, it also causes an increase in the queuing packet loss rate of the finite length queue. The system packet loss rate of the LAO method is less than that of the AMCAFQS method. Thus, the system average throughput of the LAO method is larger than that of the AMCAFQS method. The last three approaches are close to the same stability value because the signal-to-noise ratio is good. The system packet loss rate tends to the same stable value, and it can be seen from Formula (2) that the system average throughput also tends to the same stable value.

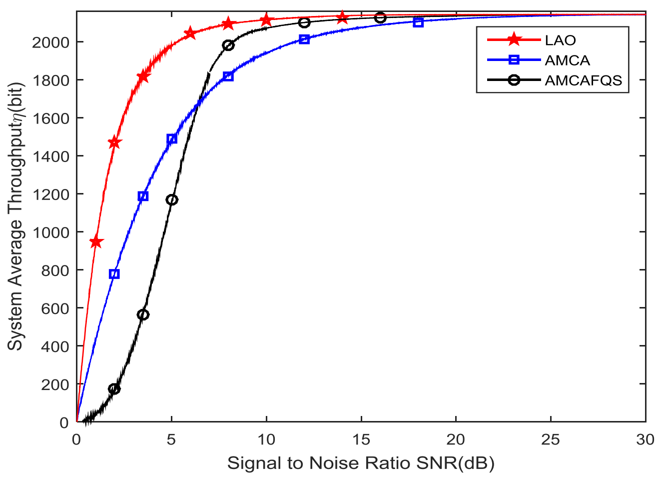

Figure 6. System throughput versus signal-to-noise ratio

6. Conclusions

In this paper, based on the characteristics of AOS space communication, a link adaptive optimization method based on the minimization of the system packet loss rate is proposed. This method combines the finite length queue and the limited retransmission of the data link layer and the adaptive modulation coding of the physical layer. The system packet loss rate is used as an objective function. By traversing the system packet loss rate corresponding to all retransmission times, the retransmission times corresponding to the minimum system packet loss rate are selected, and then the retransmission times corresponding to the minimum system packet loss rate are rounded to find the retransmission times corresponding to the suboptimal solution. The simulation shows that the proposed link adaptive optimization method based on the minimization of the system packet loss rate reduces the system packet loss rate and improves the system average throughput.

Acknowledgements

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