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Optimal Selection of Encode Rate and the Number of Frequency Slots in ISA-MAC

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Abstract

In actual system, it's a quite attractive character that one can access the channel as he wants. This paper bases on earlier ideal named ISA-MAC(Instantaneous Spectrum Availability MAC) which is promoted and listed in a paper reference [1], and focus on the optimal encode rate and the number of frequency slots of ISA-MAC. Take the probability of no-error channel transmission and the normalized throughput of the system as constrained condition, an analysis come out and be taken as a guide for reality design which is found on the mathematic model and the capability of error detection and correction, and can be adapted to CSMA, ALOHA using FH similar to ISA-MAC which is designed for fulfilling the requirement of accessing at any moment.

***Keywords**: Ad Hoc; MAC protocol; Disorder Transmission; Concatenated Coding; FH*

1. Foreword

This paper mainly analyzes the ISA-MAC mentioned in reference [1], and searches the optimal encode rate and the number of frequency slots of the ISA-MAC system with the mathematical model. The ISA-MAC includes several key techniques:

1) MIMO, which enable the message can be transmitted at any moment base on the consequence of channel statistics;
2) FH+TH, which alleviate the channel collision;
3) Error correction encode, Concatenated Coding is applied in the paper which can recover the error data caused by channel collision;
4) In this paper, MIMO will not be discussed anymore which belong to signal processing, and put the emphases on the latter two techniques.

This paper is organized as follows. In Section II, the ISA-MAC will be described briefly. In Section III, build mathematic models and make theory analysis. In Section IV, simulate and validate the raised model. Finally, submit the conclusion and summarization.

2. The Synopsis of ISA-MAC

The ISA-MAC is a kind of MPR (Multiple packet reception) MAC which has the capability of simultaneous receiving more than one packet from multiple concurrent transmissions. In 1989, it was in reference [3] that MPR was first discussed for the slotted Aloha protocol and for a random access wireless network by S. Ghez *et al*. After several proposals on channel and medium access protocols on MPR [4–6] were presented in the 90's, reference [7] was a seminal paper, the first to examine MPR, as an interaction between the physical and medium access control layers for a wireless random access network. The MPR node model used in [7] was derived from [3]. An interesting observation on the recent work on MPR is that significant results have been obtained on the capacity and throughput analysis based on the MPR model, while the enabling MPR technologies are rarely mentioned in these papers. Since 2007, a series of results on

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capacity improvement by protocol architectures that exploit MPR has been published in [8–10] .In [11], Wang *et al*. summarize their work on MPR by presenting a unifying approach for the computation of throughput capacity of wireless ad hoc networks under all data traffic patterns including unicast, multicast, broadcast, and anycast. In [12], Ghez *et al*. give two important theorems for the Slotted Aloha network with MPR. In [13], a general asymmetric MPR model is introduced and the medium-access control (MAC) capacity region is specified. From these two papers, it is clear that if the number of simultaneous transmissions is greater than the average MPR capacity, the system cannot achieve a maximum stable throughput. Dua [14] analyzed a one-dimensional Markov chain which captures the evolution of the queue of a typical transmitter in isolation in an MPR system. In [15-16], the analytical equations for the characteristics of a relay node's queue such as average queue length and stability conditions are studied for the throughput per user. There are a number of techniques which allow simultaneous decoding of packets on a receiver but in many papers MPR is said to be realized with CDMA or MIMO. In [17], the authors describe a many-to-many communication in which the transmissions are divided in frequency to allow the receiver to decode the packets. In [18], the authors present Known Modulus Algorithms (KMA) to allow packet separation in asynchronous ad hoc networks. In [19], a variation of KMA, Algebraic KMA (AKMA), is proposed based on a matrix perturbation analysis. Other signal separation techniques based on a Constant Modulus Algorithm (CMA) or Multiple Modulus Algorithm (MMA) [20] requires blind equalization and are usually not efficient for MPR. Speak of MAC with MPR, in [5] Mergen and Tong proposed a multiple-access protocol based on receiver controlled transmission (RCT). RCT is a hybrid scheduled and random access MAC, which applies scheduling to determine the receiving nodes and then the transmitters for each receiving node. RCT aims to maximize local throughput by granting an appropriate subset of users so that the varying levels of MPR capability are exploited. In [21], Zhao and Tong designed a multiqueue service room (MQSR) protocol, which exploits MPR capability to handle users with different quality of service constraints. Yu and Giannakis associate a contention tree algorithm with SIC to propose SICTA in [22]. They aim to create a collision resolution access protocol, from the packet point of view. It turns out to be an interesting approach for MPR. In [23-24], Celik *et al*. point out that the near-far problem reminds an important factor which degrades the throughput and the fairness in a wireless network even with MPR capability. To overcome this problem, they added an alternative model to the MAC protocol in which a node decreases its transmission probability following success and increases it following failure. Reference [25] proposed a dynamic multi-group priority queueing protocol to exploit the cooperative diversity for improving the system throughput. In [26], Crichigno *et al*. address the minimum length scheduling problem in one-hop multiaccess networks. They define the capacity region as the closure of the convex hull of

A set of rate vectors. Reference [27] studies the neighbor discovery algorithm by incorporate multi-packet reception capability. Jeon and Ephremides consider neighbor discovery by incorporating

MPR and Physical-Layer signal processing.

The main idea of ISA-MAC protocol is described as follows:

In the system, communication channel and quantity present diversity states. To satisfy the requirement for time-sensitive, the information is divided into several categories which takes different transmission priority refers to the degree of time sensitive. In fact, the system terminal equips with one transmission RF channel and multiple-reception communication RF channel. Prior to transmission, the system terminal checks the status of channel according to the channel statistical result from multiple-receivers which will monitor the channel during idle, and make sure if there is some margin for certain priority message to transmit, communication capability is limited by upper capability which will lead to communication collapse if it is exceeded, so only a certain amount of transmission exists at an instantaneous moment. The specific processing of transmission is shown in flow chart.

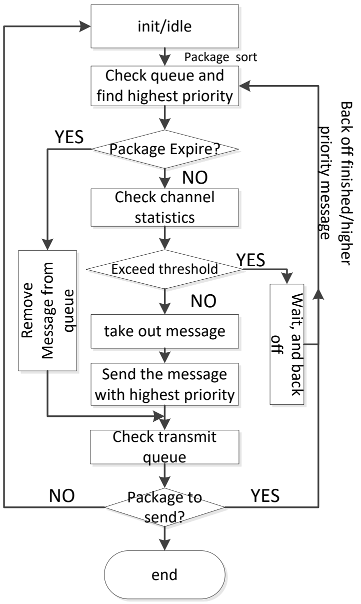

Figure 1. ISA-MAC Protocol Flow Chart

The characters of ISA-MAC include:

The message can be sent as it need only if the channel is not saturated;

By means of controlling the volume of business of channel, the total of business could keep stable level;

If the volume of business increase, it is helpful to keep the first successful delivery probability by postponing or decreasing the lower priority message;

Provide highest throughput for the higher priority message.

ISA-MAC protocol aims at the goal to satisfy system users with the high probability of first successful delivery under the appropriate network burden.

3. Mathematic Model and Theory Analysis

The communication model adapted by ISA-MAC refers to the reference [3], as a depiction in drawing 2.

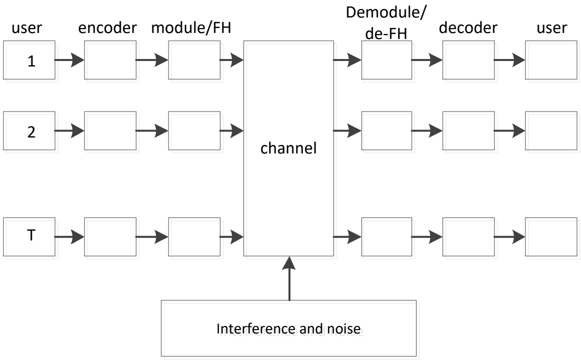

Figure 2. The Communication Model of ISA-MAC

Given that there exist T users in the system, they can easily communicate with one transmitter and multiple receivers, which is a mechanism of avoiding the failure caused by collision even if no appointment and coordination. If the message need be sent out, it will be encoded with the Concatenated Code at first, and accompany scrambling and interleaving *etc.* processing technique to depress BER. Besides, the system utilizes a combination of FH-TH spread spectrum mode to gain the two benefits: first, FH-TH could be regarded as a space diversity technology and bring the processing gain; second, FH-TH could reduce the collision probability and improve the probability of first successful delivery.

Because of FH-TH, the system modulation should adopt the constant envelope modulation technology in case of emergence of the signal leakage problem. More discussions concern the Concatenated Code could be found in reference [1], and would not be described any more.

4. The Optimal Rate of Code

Assume P presents the number of frequency slots, which means there are P parallel, contention free frequency slots at any time. But it is not hard to imagine that there will be a "collision" if multiple users occupy the same frequency point transmission. In fact, asynchronous frequency hopping is much easier to realize than synchronous frequency hopping because the Time synchronization is very difficult in a large scale scene, so asynchronous frequency hopping is adopted in reference [1] advisedly. In reference [1], the probability of collision in the asynchronous FH model is shown:

$$p_h = \frac{1}{p}(2 - \frac{1}{p})$$

Assume Q presents the number of active users, Q<T, T presents the number of total system users, which means the presence of Q instantaneous transmission at the same time, and the probability of system collision is:

$$p_{hit} = 1 - (1 - p_h)^{Q-1}$$

System deploys the Concatenated Code, in which the out code is (n, l), the inner code is (N, L), the rate of Concatenated Code is:

$$rR = lL/nN$$

In which, $r = l/n$, $R = L/N$ presents the rate of the out code and the rate of the inner code respectively.

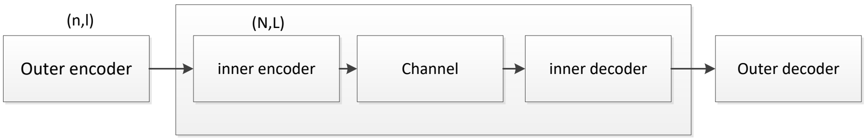

Figure 3. Schematic Diagram of Concatenated Coding

To simplify the analysis condition, each hop contains an inner symbol, which is composed of outer code separated after scrambling. In terms of channel, it’s necessary to analyze the inner code, and introduce the characteristics of the out code in the process of analysis. Set e as the inner error correcting capability, f represents the inner Error detection capability ($e \leq f$), and meet the conditions: $e + f < d_{min}$, d_{min} is the minimum hamming distance of the inner code. When errors occur during transmission, if detected, then the symbol will be erased, however, if the errors are not detected, nor corrected, then the inner code will output error symbol. Set p_d and p_{ud} as the error detection probability and the error detection probability respectively, the out code have the capability to correct and erasure the output error symbol from the inner code.

In this paper, using the $RS-RS$ concatenated code, and the RS code have the error correction capability is t bits, and the Error detection capability is e bits, $e+2t\le n-l$.In the memoryless channel, the probability of decoding as follows:

$$P_c(I)=\sum_{k+2m\le n-l}\binom{n}{k,m}p_d^k p_{ud}^m(1-p_d-p_{ud})^{n-k-m}$$

In which,

$$\binom{n}{k,m}\square\frac{n!}{k!m!(\mathrm{n}-k-m)!}$$

Build the mathematic model, take the sample X_i $(i=1,2,\cdots,n)$

$$X_i=\begin{cases}0, & \text{under correct reception}\\ 1, & \text{if symbol is erased}\\ 2, & \text{if symbol is not correct}\end{cases}$$

Set

$$Y_n\square\sum_{i=1}^{n}X_i$$

$$Z_n\square\frac{Y_n-E(Y_n)}{\sqrt{Var(Y_n)}}$$

$E(Y_n)$ and $Var(Y_n)$ represent the mean and variance of Y_n respectively, Y_n is the sum of the erased characters and two times the error character. To decode correctly, meet the following results:

$$P_c(I)=P(Y_n\le n-l)$$

Because the sample X_i $(i=1,2,\cdots,n)$ is independent to each other, according to the central limit theory

$$\lim_{n\to\infty}P(Z_n\le \mathrm{z})=\int_{-\infty}^{x}\frac{1}{\sqrt{2\pi}}e^{-u^2/2}du$$

Besides,

$$E(Y_n)=n(p_d+2p_{ud})$$

$$Var(Y_n)=n(p_d-p_d^2+4p_{ud}-4p_{ud}^2-4p_dp_{ud})$$

In which

$$P(X_i=1)=p_d$$

$$P(X_i=2)=p_{ud}$$

When l tends to infinite, and $r=l/n$ tends to be constant, the conclusion as follows:

$$\begin{aligned}P_c(I)&=P(Y_n\le n-l)\\ &=P(Z_n\le\frac{n-l-E(Y_n)}{\sqrt{Var(Y_n)}})\\ &=P\left(Z_n\le\frac{\sqrt{n}(1-r-p_d-2p_{ud})}{\sqrt{p_d-p_d^2+4p_{ud}-4p_{ud}^2-4p_dp_{ud}}}\right)\\ &=\begin{cases}1,1-r-p_d-2p_{ud}>0\\ 0.5,1-r-p_d-2p_{ud}=0 \quad \text{(according to the characteristics of Normal distribution)}\\ 0,1-r-p_d-2p_{ud}<0\end{cases}\end{aligned}$$

Obviously, to realize the communication transmission error free, the condition must be satisfied:

$$r<1-p_d-2p_{ud}$$

The above results only for the derivation aim at the outer code. To achieve the constraints on the overall concatenated code, the inner code rate selected can be brought into the above results and obtain the following results:

$$rR < \frac{L}{N}(1 - p_d - 2p_{ud})$$

If n is limited, According to the above derivation, the result is as follows:

$$P_c(I) \approx F\left(Z_n \le \frac{\sqrt{n}(1 - r - p_d - 2p_{ud})}{\sqrt{p_d - p_d^2 + 4p_{ud} - 4p_{ud}^2 - 4p_d p_{ud}}} \right)$$

Correspondingly, the constraint to the out code rate could be derived under the condition that the code type and the quantity of simultaneous transmission user are certain.

$$r < 1 - p_d - 2p_{ud} - \alpha\sqrt{(p_d - p_d^2 + 4p_{ud} - 4p_{ud}^2 - 4p_d p_{ud})/n}$$

In which

$$\alpha = F^{-1}(1 - \hat{P}_E)$$

Among it, $F_x = \int_{-\infty}^{x} (2\pi)^{\frac{-1}{2}} e^{-u^2/2}\, du$ and $\widehat{P_E}$ is the probability of error threshold which could be Approximated by $P_E = 1 - P_c(I) \le \widehat{P_E}$. Based on the above analysis, the overall rate of concatenated code constraint formulas is available:

$$rR \le \frac{L}{N}\left[1 - p_d - 2p_{ud} - \alpha\sqrt{(p_d - p_d^2 + 4p_{ud} - 4p_{ud}^2 - 4p_d p_{ud})/n}\right]$$

In fact, in term of the channel model, the decoding ability of the out code is actually constrained by the decoding ability the inner code. To make sure the $RS - RS$ Concatenated Code can work well, it's necessary to consider the two levels encoding comprehensively, in other words, the inner code decoding output can be corrected or detected by the out code is the precondition that assure p_d and p_{ud} meet the demand of the system.

In system design, if the out code employs short code, and the inner code employs the identical length or longer code, the inner code will be abandoned straightly if the error output of the inner code exceeds the error detection/correction capability itself which lead to lose the advantage of concatenated code instead. Therefore, the outer code using long code, and contains $M = [L/n]$ messages in each symbol, if the scrambling and interleaving processing are employed in information process, the error information will be uniformly distributed in M messages, and q bits error character in each message whose length is $n = L/M$ bits. Set the error detecting capability of inner code is S bits, and the error correcting capability is D bits. To achieve successful decoding, the error number from inner code decoding output must be not greater than the error detection capability of out code.

$$Mq \le n - l \,\&\, q \le (N - L)/2$$

The reason why $q \le (N - L)/2$ have to be met is the decoding ability of out code will directly fail or information cannot be decoded correctly if the error number from inner code decoding output beyond the error correcting capability. The range of q: $U = min[((n-1)/M), (N-L)/2]$, p_d represents the probability of successful decoding, p_{ud} represents the probability of failing to detect error.

$$\begin{aligned} p_d' &= p(q \le U) \\ &= \sum_{i=0}^{U} p(N_{ERROR} = i) \\ &= \sum_{i=0}^{U} \left[p(N_{ERROR} = i / H) p_{hit} + p(N_{ERROR} = i / \overline{H})(1 - p_{hit}) \right] \end{aligned}$$

In which, $p(N_{ERROR} = i/H)$ presents the probability of error under collision, $p(N_{ERROR} = i/\overline{H})$ presents the probability of error under no collision.

$$\begin{aligned} & p(N_{ERROR} = i / H) \\ &= \sum_{i=1}^{Q-1} \binom{Q-1}{i} p(N_{ERROR} = i / H(Q)) p_{hit}'^{\,i} (1 - p_{hit}'^{\,i})^{Q-1} \end{aligned}$$

And

$$\begin{aligned} & p(N_{ERROR} = i / H(Q)) \\ &= \sum_{i=1}^{n} p(N_{ERROR} = i / N_{hit} = s) p(N_{hit} = s / H(Q)) \end{aligned}$$

$$\begin{aligned} & p(N_{ERROR} = i / N_{hit} = s) \\ &= \left(\frac{1}{2}\right)^{s} \sum_{z=0}^{i} \binom{s}{z} \binom{n-s}{i-z} p_0^{\,i-z} (1 - p_0)^{n-s-i+z} \end{aligned}$$

$$\begin{aligned} & p(N_{hit} = s / H(Q)) \\ &= \begin{cases} 2\left(\dfrac{1}{2n}\right)^{Q}, & s = 1 \\ \dfrac{3s^{Q} - 4(s-1)^{Q} + (s-2)^{Q}}{2n^{Q}}, & 2 \le s \le n-1 \\ 1 - \dfrac{2 - \sum_{i=1}^{n-1} 3i^{Q} - 4(i-1)^{Q} + (i-2)^{Q}}{2n^{Q}}, & s = n \end{cases} \end{aligned}$$

Besides,

$$p(N_{ERROR} = i / \overline{H})(1 - p_{hit}) = \binom{n}{t} p_0^{t} (1 - p_0^{t})^{n-t} (1 - p_{hit})$$

p_0 represents the BER, and equals to 10^{-6}.

Suppose there is no correlation between characters in inner symbol, so

$$p_d = p_d'^{\,M}$$

Similarly

$$p_{ud}' = p(q > n - l) = \sum_{i=n-l}^{n} p(N_{ERROR} = i)$$

And

$$p_{ud} = \sum_{i=1}^{M} \binom{M}{i} p_{ud}'^{\,i} (1 - p_{ud}')^{M-i}$$

Which can be calculated directly on the basis of the derived result?

According to reference [2], the general throughput can be defined as

$$W = \frac{r' K P_C}{q}$$

In which, r' is the rate of Concatenated Code, K is the quantity of active users, P_C is the probability of no error transmission, q is the quantity of frequency slots. The derivation processing is not noticed any more; take the related parameters Into the Mentioned equation.

$$W = \frac{l}{n} \bullet \frac{L}{N} \bullet \frac{KP_C}{q}$$

Set the rate of the out code as a constant, therefore

$$\lim_{n \to \infty} W = \frac{L}{N} \bullet \frac{K(1 - p_d - p_{ud})}{q}$$

Which gains the appropriate quantity of frequency slots by approaching maximum network throughput.

4. The Simulation and Verification

The protocol is analyzed and stimulated with MATLAB, and the statistical results are as follows:

Table 1. The Statistical Results for Optimal Rate

Serial number	Inner code	Out code	p_d	p_{ud}	α	rR
1	RS (31,15)	RS (255,239)	0.0014	1.3574e-81	3.9	0.9373
2	RS (31,15)	RS (255,127)	0.0165	8.4839e-82	3.9	0.4980
3	RS (15,7)	RS (255,127)	0.00061	1.2995e-38	3.9	0.4980
4	RS (31,15)	RS (511,255)	0.00061	1.5271e-81	3.9	0.4990
5	RS (31,15)	RS (1023,511)	8.7261e-07	2.8845e-81	3.9	0.4995
6	RS (31,15)	RS (2047, 1023)	1.7301e-12	5.5994e-81	3.9	0.4998
7	RS (15,7)	RS (2047, 1023)	2.5519e-25	9.9631e-38	3.9	0.4998

According to the above analysis, $\alpha = F^{-1}(1 - P_E)$ and $P_E = 1 - P_c(I) \leq \widetilde{P_E}$, to realize no error communication, $P_c(I)$ $P_c(I)$ have to be 1 that led P_E to be 0, α take value 3.9 by looking up the standard normal distribution table.

The content of Table 1 is part of the testing data which provide the consult as follows:

1. the rate of system is effected by the rate of out code mainly, if the rate of out code is larger, the rate of system will be larger which imply higher code efficiency and poorer error correction and detection capability;
2. When the rate of out code reach certain level, the rate of system will trend to be stable, it's proved to be limited that improve the system by increasing the rate of out code.
3. If the rate of inner code and the rate of out code are all tend to be 0.5. The rate of system will close to be 0.5, but never be 0.5. It's the proper combination to ensure the system gain best error correction capability and error detection capability.
4. In the algorithm, information is encoded by super long code before being divided into short information which is encoded by short code. Through the way, system gains the stronger decoding capability to recover system information even if a large number of short codes improve the probability of collision.
5. It's easy to choose the optimal combination of code: RS (31, 15) as inner code, RS (255,127) as out code for two reasons: first, p_d is relatively high; second, the efficiency of encoding is high.

The analysis for frequency slot can be viewed in Table 2.

Table 2. Simulation for Frequency Slot

Serial number	L/N	K	P_d	P_{ud}	q	W
1	1/6	15	0.0014	1.3574e-81	20	0.1248
2	1/5	15	0.0165	8.4839e-82	20	0.1475
3	1/4	15	0.00061	1.2995e-38	20	0.1874
4	1/3	15	0.00061	1.5271e-81	20	0.2498
5	1/2	15	8.7261e-07	2.8845e-81	20	0.3750
6	1/2	15	1.7301e-12	5.5994e-81	20	0.3750
7	1/2	15	2.5519e-25	9.9631e-38	20	0.3750
8	1/2	15	0.0014	1.3574e-81	20	0.3745
9	1/2	15	0.0165	8.4839e-82	20	0.3688
10	1/2	15	0.00061	1.2995e-38	20	0.3748
11	1/2	15	0.00061	1.5271e-81	20	0.3748
12	1/2	15	8.7261e-07	2.8845e-81	20	0.3750
13	1/2	15	1.7301e-12	5.5994e-81	20	0.3750
14	1/2	15	2.5519e-25	9.9631e-38	20	0.3750
15	1/2	15	0.0014	1.3574e-81	18	0.4161
16	1/2	15	0.0165	8.4839e-82	15	0.4918
17	1/2	15	0.00061	1.2995e-38	13	0.5766
18	1/2	15	0.00061	1.5271e-81	10	0.7495
19	1/2	15	8.7261e-07	2.8845e-81	8	0.9375
20	1/2	15	1.7301e-12	5.5994e-81	5	1.5
21	1/2	15	0.0165	8.4839e-82	20	0.3688
22	1/2	15	0.0165	8.4839e-82	15	0.4918
23	1/2	15	0.0165	8.4839e-82	13	0.5674
24	1/2	15	0.0165	8.4839e-82	10	0.7376
25	1/2	15	0.0165	8.4839e-82	8	0.9220
26	1/2	15	0.0165	8.4839e-82	5	1.4753

The conclusion could be summarized:

1. Promoting the rate of out code is helpful to improve the throughput of systems;
2. It works to reduce q to improve the throughput of system when the rate of out code is fixed to some extent over which the rule will be invalid, for example $W>1$(W is a normalized parameters and can't exceed 1).It have been proved in the existing literature that the threshold of normalized throughput of async-FH is e^{-1}, and choose the value of q to achieve the throughput close to e^{-1}, such as 20,15 or 13.

5. Conclusion

The paper focus on the analysis for searching the optimal rate and frequency slot by mathematic model which is validated with Matlab that the theoretical analysis could be utilized to optimize the code type of $RS-RS$ Concatenated Code and the frequency slot.

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