Voice over Internet Protocol over IEEE 802.11 Wireless Local Area Networks and effective admission control with transmission interval adaptation

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Abstract: Voice over Internet Protocol (VoIP) over Wireless Local Area Networks (WLANs) is an important application of WLANs and it has gained more and more attention recently. In this article, we analyse the maximum number of VoIP calls supportable in a WLAN and propose a new call admission control strategy, namely, Adaptive Transmission Interval Call Admission Control (ATICAC) to enhance VoIP calls in 802.11 WLANs. In ATICAC strategy, Base Station adaptively changes transmission interval of the active stations to prevent the network from saturation by controlling the average collision probability of the network. The proposed model gives a very good agreement with our simulation results. ATICAC can not only ensure the Quality of Service (QoS) of VoIP calls in 802.11 WLANs in the network, but also increase the number of VoIP calls in the network.

Keywords: call admission control; IEEE 802.11; Quality of Service; QoS; Voice over Internet Protocol; VoIP; Voice over Wireless Local Area Networks; VoWLANs; Wireless Local Area Networks; WLANs.

Reference to this paper should be made as follows: Chen, Z., Wang, L., Wang, X. and Chen, H-H. (2008) 'Voice over Internet Protocol over IEEE 802.11 Wireless Local Area Networks and effective admission control with transmission interval adaptation', *Int. J. Autonomous and Adaptive Communications Systems*, Vol. 1, No. 1, pp.82–105.

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1 Introduction

Voice over Internet Protocol **(**VoIP) has become more and more popular recently for its simplified infrastructure and significant cost reduction. Using VoIP over the networks, there is no need to separate cabling for telephone system and the phone can be taken when offices move with the same telephone number. It is because a VoIP exchange is based on software rather than hardware. It is easier to alter as well as maintain with operating costs lowered by 20–30% (Intel Co., Ltd, 2007).

With the diminishing cost of electronic hardware, IEEE 802.11-based Wireless Local Area Networks (WLANs) have been massively deployed in public and residential places such as classrooms, airports and apartments, and more and more devices and peripherals are integrated with WLAN access capability. Due to these important developments in recent years, there have been greatly increasing interests in VoIP in WLANs, in which the IEEE 802.11 Distributed Coordination Function (DCF) protocol or enhanced DCF protocol is used. However, many challenges still remain in Voice over WLANs (VoWLANs). It is well-known that widely deployed IEEE 802.11 WLANs employ a contention-based Medium Access Control (MAC) protocol as the DCF. Although DCF can well support the best effort traffic, it may introduce arbitrarily long delay and delay jitters. Thus, it is unsuitable for real-time applications with strict Quality of Services (QoS) requirements. In addition, unlike cellular networks where dedicated channels are assigned to voice traffic, voice packets in WLANs are multiplexed with data traffic. When data traffic load increases, the QoS of VoWLAN would be severely degraded. It is a challenging task to provide QoS for voice traffic while maintaining as high throughput as possible for data traffic. In fact, many issues of how the IEEE 802.11 WLAN can support QoS of VoIP have not yet been well understood.

In this article, Section 2 will discuss the related works reported in the literature. We endeavour to address these issues through both analysis and simulations in Section 3. We develop Adaptive Transmission Interval Call Admission Control (ATICAC) strategy to ensure QoS of VoIP in IEEE 802.11 WLANs for supporting major QoS of VoIP metrics, i.e. throughput, delay and packet loss rate. Our ATICAC strategy can also support more calls in the network compared with normal call admission control schemes given in Section 4. Our simulations given in Section 5 demonstrate the effectiveness of our analytical model and show the performance of ATICAC strategy.

2 Related works

Bianchi used a Markov process to model DCF and evaluated the channel throughput, and frame loss as a function of the number of wireless stations in Bianchi (2000). Initial studies on the performance of real-time applications over 802.11 were presented by authors in Garg and Kappes (2003) and Zhai, Chen and Fang (2004). The authors studied the inherent limitations of the 802.11 a/b DCF in supporting VoIP calls over a WLAN in Shin and Schulzrinne (2007).

Current research works (Veres et al., 2001; Li et al., 2007) as well as references therein and the Enhanced DCF (EDCF) defined in the IEEE 802.11e draft (Choi et al., 2003; IEEE 802 standards and the committee, 2004) tend to provide differentiated service rather than stringent QoS assurance. Analysis of interference model in wireless mesh network was made by Wei (2006) and the authors also proposed a call admission control scheme with interference capacity. Several performance optimisation schemes were proposed for WLANs to improve the VoIP quality such as those reported in Yu, Choi and Lee (2004), Wang, Liew and Li (2005) and Li, Li and Cai (2006). Authors in Yu, Choi and Lee (2004) proposed the use of dual queue of 802.11 MAC to provide priority to VoIP; while (Wang, Liew and Li, 2005) proposed packet aggregation to increase capacity and (Li, Li and Cai, 2006) proposed an adaptive transmission algorithm over an IEEE 802.11 WLAN that supports integrated voice and data services, where data traffic is transmitted with DCF, and voice transmission is carried out with Point Coordination Function (PCF).

3 Analytical study of Voice over Internet Protocol in the IEEE 802.11 Wireless Local Area Networks

This section is focused on the analysis of the performance of VoIP in the IEEE 802.11 DCF. It is noted that in the following analysis, the hidden terminal problem is ignored for analysis simplicity. This is because in a typical WLAN environment, every node can sense all the other transmissions, although it may not necessarily be able to correctly receive the packets from all other nodes (Zhai, Chen and Fang, 2005).

3.1 Maximum number of Voice over Internet Protocol calls

Unlike a wired network, the actual available bandwidth B_{avl} in a wireless network is usually less than the network average bandwidth B_{avg} due to the collision and backoff idle. If we define T_{suc} as the average time period associated with successful transmission, T_{col} as the average time period associated with packet collision and T_{idle} as the average time period associated with backoff idle in a certain time interval, we can obtain

$$B_{\text{avl}} = \frac{T_{\text{suc}}}{T_{\text{suc}} + T_{\text{col}} + T_{\text{idle}}} \times B_{\text{avg}}. \tag{1}$$

To calculate T_{suc}, T_{col} and T_{idle}, we assume that the total active number of full duplex VoIP calls in the network is n. To simplify the analysis and yet reveal the characteristics of the VoIP in IEEE 802.11 MAC protocol, we assume that one full duplex VoIP call equals two half duplex connections established between active stations and Base Station (BS). Therefore, we can presume that the VoIP traffic is uniformly distributed among these $2n$ active stations. If we assume that the transmission probability for each active station in any time slot is τ, we can obtain the following equations according to IEEE 802 standards and the committee (1999) and Bianchi (2000) as

$$\begin{cases} p_{\text{i}} = (1-\tau)^{2n} \\ p_{\text{s}} = 2n\tau(1-\tau)^{2n-1} \\ p_{\text{c}} = 1 - p_{\text{i}} - p_{\text{s}} = 1-(1-\tau)^{2n} - 2np(1-\tau)^{2n-1} \end{cases} \tag{2}$$

where p_{i} is the probability that the observed backoff time slot is idle, p_{s} is the probability that there is one successful transmission and p_{c} is the collision probability that there are at least two concurrent transmissions at the same backoff time slot. Hence, Equation (1) becomes

$$B_{\text{avl}} = \frac{p_{\text{s}}T_{\text{s}}}{p_{\text{s}}T_{\text{s}} + p_{\text{c}}T_{\text{c}} + p_{\text{i}}T_{\text{i}}} \times B_{\text{avg}} \tag{3}$$

where T_{s} is the time of a successful transmission, T_{c} is the time wasted by a packet collision and T_{i} is the duration of an empty slot time. We know from Bianchi (2000)

$$T_{\text{s}} = DIFS + DATA + SIFS + ACK \tag{4}$$

$$T_{\text{c}} = DATA + EIFS \tag{5}$$

$$T_{\text{i}} = \text{A Slot Time} \tag{6}$$

where DATA is the time needed to transmit a data packet including IP header and MAC header, and all of these values are shown in Table 1.

Here, we do not use RTS/CTS mechanism, because the RTS and CTS frames are too large as shown in Table 1 if compared with the payload data. Therefore, using RTS/CTS mechanism will waste certain bandwidth.

The payload rate of voice data for each station is equal to R_{codec}, which represents the bit rate of codec. Although the R_{codec} is very low (for example, R_{codec} of Codec G.729a, which is the mostly used codec in VoIP applications, is 8 kbps), the bandwidth required to transmit these data payload is very large. It is because that if compared with the time period T_{s} which represents the time needed to send a packet successfully, the time to transmit the payload information T_{p} is much shorter. As payload information occupies only a small part of data packet due to the overhead added in each layer as shown in Table 1, and from Equation (4), we know that the time to send data is just a part of T_{s}. Therefore, we can obtain the required bandwidth as

$$B_{\text{req}} = \frac{T_{\text{s}}}{T_{\text{p}}} \times R_{\text{codec}} \tag{7}$$

Table 1 IEEE 802.11 system parameters

R_{data} Bit rate for DATA packets	2 Mbps
R_{basic} Bit rate for RTS/CTS/ACK	2 Mbps
CW_{min}	31
CW_{max}	1,024
PLCPDataRate	1 Mbps
A Slot Time	20 µs
SIFS	10 µs
DIFS	50 µs
EIFS	364 µs
PreambleLength	144 bits
PLCPHeaderLength	48 bits
MAC header	224 bits
IP header	160 bits
DATA packet	Payload size + MAC header + IP header = Payload size + 384 bits
RTS	160 bits
CTS and ACK	112 bits

where T_p can be calculated by

$$T_p = \text{Frame Size} \times \text{Frames Per Packet} / R_{data} \tag{8}$$

where R_{data} is the data rate of network shown in Table 1, and Frame Size is the length of a frame coded by codec. For example, the frame size of codec G.729a is 80 bits (International Telecommunication Union Telecommunication Standardisation Sector, 2007). Thus, we can obtain the maximum number of VoIP calls N as

$$N = \frac{B_{avl}}{(2 \times B_{req})} \tag{9}$$

where the factor '2' indicates that VoIP is a dual connections.

Bianchi (2000) provides us a group of equations to calculate τ in Equation (2) for a saturate network. In our cases, these equations become

$$\tau = \frac{2(1-2p)}{(1-2p)(W+1)+pW(1-(2p)^m)} \tag{10}$$

$$p = 1-(1-\tau)^{2n-1} \tag{11}$$

where p is referred to as conditional collision probability which represents the probability that a collision occurs if a packet starts transmitting over the channel, W is equal to CW_{min}, CW_{max} is equal to $2^m W$, and n is the number of VoIP calls in the network. If we have τ, we can use Equations (2) and (3) to calculate B_{avl} of network when it comes into saturate mode. As shown in Zhai, Chen and Fang (2005), the B_{avl} of a network in the saturate status is 0.9 times of the maximum B_{avl} a network can provide. Therefore, if we use Equations (10) and (11) to calculate B_{avl}, the Equation (9) should be modified into

$$N = \frac{B_{\text{avl}}/0.9}{2 \times B_{\text{req}}}. \tag{12}$$

We can simplify N as

$$N = \frac{p_s T_p}{p_s T_s + p_c T_c + p_i T_i} \times \frac{B_{\text{avg}} / 0.9}{R_{\text{codec}} \times 2}. \tag{13}$$

Thus, given the parameters of the codec and the network and if we let n equals to N, we can obtain the maximum number of calls network can support by solving Equations (2), (10), (11) and (13).

Take Codec G.729a and network parameters shown in Table 1 as examples. For Codec G.729a, the bit rate of codec R_{codec} is 8 kbps (80 bits for 10 ms frames). If we choose two frames per packet to transmit, which represents 20 ms interval to transmit 20 bytes. Using Equations (4)–(6), we can obtain that $T_s = 50 + 464 + 10 + 248 = 772$ μs, $T_c = 464 + 364 = 828$ μs, $T_p = 80$ μs, and $T_i = 20$ μs. Then, we can obtain the maximum number of calls network can support N is 10.4945.

Table 2 summaries the maximum number of VoIP calls for different number of frames per packet. In these calculations, network parameters are shown in Table 1 and we use codec G.729a. We can see from Table 1 that the larger number of frames per packet is, the more VoIP calls are allowed in the network since the required bandwidth is lower; while larger number of frames per packet means larger delay, for example, four frames per packet means 40 ms more codec delays. If the number of frames per packet is too large, it will not satisfy the quality of VoIP showed in the next Section 3.2.

Table 2 Maximum number of Voice over Internet Protocol calls for G.729A

Frames/packet	*Required bandwidth (kbps)*	*Number of calls*
1	144.4	5.9251
2	76.2	10.4945
3	53.47	14.776
4	42.1	17.9248
5	35.28	20.9946
6	30.73	23.7042
7	27.48	26.1102
8	25.05	28.4005
9	23.15	30.4697
10	21.68	32.3451

3.2 Voice quality measures

Before obtaining the maximum number of VoIP calls experimentally, we should define the measurement of voice quality in engineering. We use a metric proposed in Cle and Rosenbluth (2001) which takes into account mouth to ear delay, loss rate and types of the encoder. The quality of voice is defined by the R-score

$$R = 94.2 - 0.024d - 11 - 40\log(1+10e) - 0.11(d-177.3)H(d-177.3) \qquad (14)$$

where the notations can be explained as follows:

1. $d = d_{\text{codec}} + d_{\text{jitterbuffer}} + d_{\text{network}}$ is the total mouth to ear delay comprising vocoder delay which includes 5 ms look-ahead delay, 10 ms coding delay per frame, delay in the de-jitter buffer and network delay. The network delay includes delays in sending buffer and retransmissions.
2. $e = e_{\text{network}} + (1 - e_{\text{network}})e_{\text{jitter}}$ is the total loss including network and jitter losses.
3. $H(x)$ is the Heaviside function defined as

   $$H(x) = \begin{cases} 1 & \text{if } x > 0 \\ 0 & \text{otherwise.} \end{cases}$$

4. the parameters used are specific to the G.729a encoder with uniformly distributed loss.

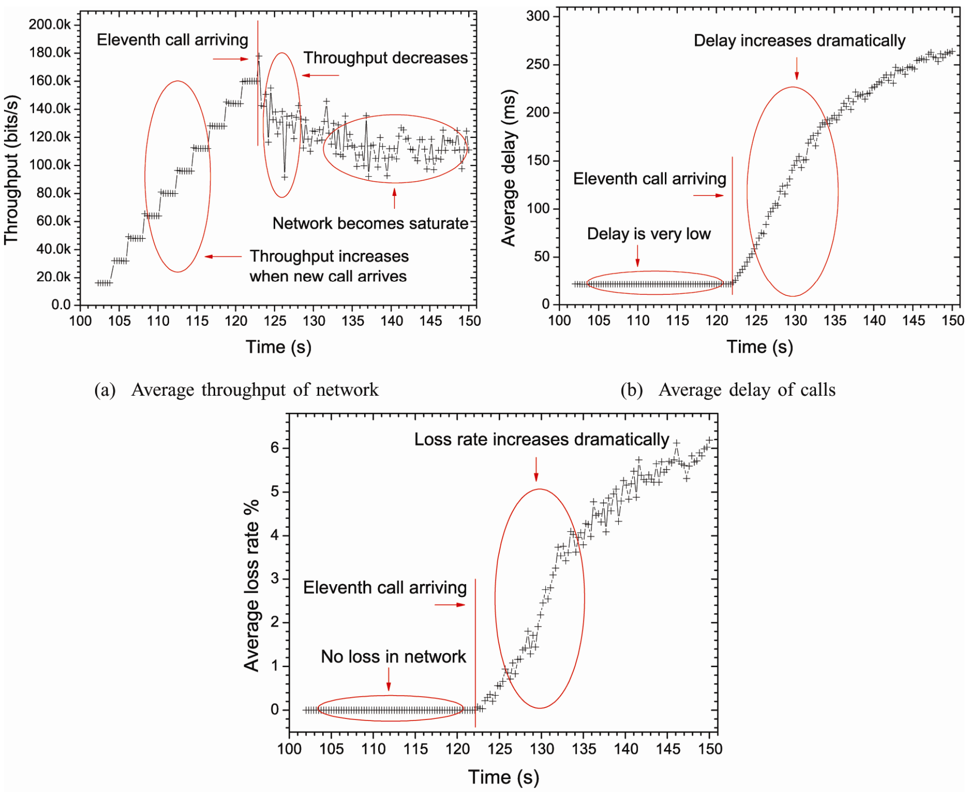
The **R**-score greater than 70 is considered as acceptable quality.We can calculate from Equation (14) that the total delay should be no larger than 244 ms, which means the delay in the network should be less than 220 ms even there is no loss in the network and jitter. In general situation, to ensure the quality of VoIP in WLANs, the network delay should not be larger than 150 ms and the network loss rate should be less than 3%.

3.3 Maximum number of Voice over Internet Protocol connections

In this section, we will show the maximum number of VoIP calls in ns-2 (NS-2 Simulator, 2007) simulations with the network parameters shown in Table 1 and Codec 729.a with two frames per packet, which means 20 bytes per 20 ms. Assume that the VoIP calls join the network one by one every two seconds. We can see from Figure 1 that when the 11th call joins the network, the quality of all VoIP calls in the network becomes bad, and the throughput of the network drops slightly. It indicates that the maximum number of VoIP calls is 10, which accurately matches the value N given in Section 3.1.

To identify the reason why the 11th call makes the network into poor situation, we plot Figure 2 to analyse the average collision probability of the network and the average idle time of the network. We find that before the 11th call comes, the average collision probability p_c increases slightly from 0 to 0.03. The average idle time drops quickly at the beginning because when the number of calls increases, the idle time due to no packet to be sent decreases quickly. The average idle time drops slightly when there are more calls in the network, since at that time most of network idle time is due to backoff process and DIFS (DCF Interframe Space) time. When the 11th call comes into the network, the average collision probability p_c increases quickly to 0.1 from 0.03 and then jumps to 0.23, and the average network idle time decreases from 200 to 100 µs slightly. This means in Equation (1) the parameter T_{col} becomes much larger and the average system idle time T_{idle} becomes smaller. Thus, the available bandwidth B_{avl} becomes lower.

Figure 1 Call quality statistics with the number of calls increase (see online version for colours)



(a) Average throughput of network

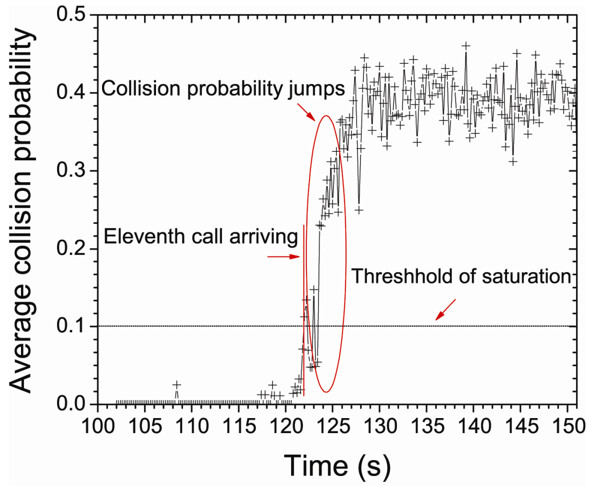
(b) Average delay of calls

(c) Average loss rate of calls

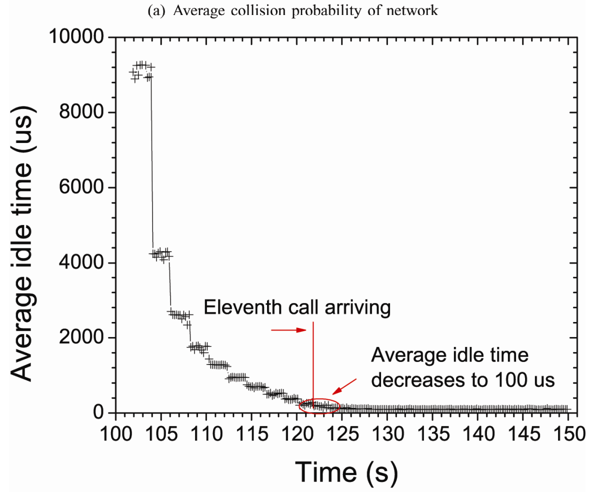
Therefore, we can conclude that the high collision probability takes the main responsibility that the average quality of VoIP call becomes poor when the 11th call comes.

We know from Zhai, Chen and Fang (2004) that if the average collision probability of the network p_c is larger than p'_c which is around 0.1, the network comes into a saturate status. In the saturate network, the average MAC service time is longer than the packet arrival interval. Thus, the queue will always have packets to send and the delay in the queue will be accumulated. Therefore, no call is in good quality if the network becomes saturate. If we want to maintain the average quality of ongoing calls, we should take some admission control strategies to ensure that the network does not come into saturation. Also, to support more VoIP calls, we should take some strategies to reduce the collisions in the network like adaptively changing the transmission interval.

Figure 2 Average collision probability and idle time of the network (see online version for colours)



(a) Average collision probability of network



(b) Average idle time of network

4 Adaptive Transmission Interval Call Admission Control strategy

In this section, we will introduce the ATICAC strategy to enhance the quality of VoIP calls in the network, aiming at preventing the network from saturation. The basic idea of ATICAC is that we find that increasing the transmission interval of existing calls can decrease the network average collision probability p_c, which determines whether the network is saturate. In the meanwhile, a large transmission interval may lead to a long delay for the voice. Thus, ATICAC is to adaptively choose the transmission interval according to the network status.

As we have discussed in the above section, if the network works in non-saturate status, the quality of exiting VoIP in the network can be ensured. The main strategy of our call admission control is to prevent the network from saturation. Using our call admission control strategy, a new call will be accepted without greatly reducing the quality of existing calls and the quality of existing calls will be improved when a call leaves.

As we have discussed in the earlier section, if we can control the average collision probability p_c of the network to keep it less than p'_c, all the quality of existing calls will be guaranteed, which is the most important objective of our call admission control strategy. Here, comes another problem that the average collision probability p_c of the network is difficult to obtain because every station in the network including BS does not know whether other nodes are in collision. Therefore, we use a new parameter called channel busyness probability as given in Zhai, Chen and Fang (2004) p_b instead of p_c, which is the probability that a node senses the channel is busy. p_b can be easily obtained by every station, since the IEEE 802.11 is a CSMA-based MAC protocol, working on the physical and virtual carrier sensing mechanisms. We will prove below that p_b is related to p_c and BS can use p_b to obtain p_c which determines the state of the network.

Given the transmission probability for each active station in any time slot τ as well as the Equation (2) we have discussed above, the channel busyness probability p_b can be expressed as

$$p_b = 1 - p_i = 1 - (1-\tau)^{2n} \tag{15}$$

where the number of connecting calls n can be known by BS. Thus, combined Equations (2) and (15), p_c can be expressed by p_b as

$$\begin{cases} p_c = 1 - (1-p_b) - 2n\tau \dfrac{1-p_b}{1-\tau} \\ \tau = 1 - \sqrt[2n]{1-p_b} \end{cases}. \tag{16}$$

Hence, BS can quickly obtain p_c through p_b and n to administrate the network.

In Table 2, we can see that different number of frames per packet, which means different transmission intervals, determines different maximum number of VoIP calls with the same network parameters. We can obtain from Equation (7) that the longer transmission interval, the less required bandwidth B_{req} is since the larger packet means the smaller overhead ratio, and less B_{req} means more calls can be allowed in the network from Equation (9). But this is not the main reason that enlarges the number of calls in the network. A longer transmitting interval means fewer packets to transmit in the network, resulting in less collisions in the network. Thus, the collision probability of the network p_c will be reduced and more VoIP calls can be allowed to come into the network.

The longer transmission interval is, the better the performance will be due to the less required bandwidth and smaller collision probability in the network, while it is not suitable for VoIP transactions since end-to-end delay of VoIP call is tightly restricted. We have known from Section 3.2 that even there is no collision in the network, the end-to-end delay should be less than 244 ms. If two users in separate WLANs build up a VoIP call, the end-to-end delay between stations and BS should be less than 122 ms without considering wired internet delay. A longer transmission interval means a larger voice delay which is a part of end-to-end delay. Therefore, there is a trade off between the number of VoIP calls and end-to-end delay, i.e. the quality of VoIP calls.

Hence, we can describe our admission control strategy with adaptive transmission intervals. In the system, we assume that the maximum system capacity is C and the maximum system capacity for data traffic is αC, where is α between zero and one. When there is a new voice call arriving at the BS, the BS should calculate the network average collision probability p_c through p_b. If p_c is smaller than p'_c, it means the network is not saturate, and the new call will be accepted; If not, it means that the network comes into saturation, and BS will decide whether to decrease the quality of existing calls by enlarging the transmission interval of all calls including the new one. If the quality of existing calls is poor, i.e. the transmission interval is 50 ms, the new call should be rejected by BS; otherwise BS will enlarge the transmission interval of all calls including the new one to accept the new call. When there is a new data traffic arriving at the BS, the BS should calculate the network average collision probability p_c through p_b. If p_c is smaller than p'_c and the existing data traffic capacity is smaller than the maximum capacity, the BS will admit this data traffic and update the existing data traffic capacity. In other case, when an existing traffic finishes, the BS will attempt to diminish the transmission interval of the rest existing calls to enhance the quality of these calls, and then calculate p_c again. If p_c is smaller than p'_c, the attempt is successful, and all the existing calls will have better quality; If not, the attempt is failed, all the existing calls will operate in previous transmission interval, and the quality of calls will not be enhanced much. The pseudocode of the admission control procedures is displayed in Figure 3.

5 Performance analysis of Adaptive Transmission Interval Call Admission Control strategy

In this section, we use two-dimensional Markov chain theory to analyse the performance of ATICAC strategy. Figure 4 indicates the transition diagram for the system not using ATICAC strategy and Figure 5 shows the transition diagram for the system using ATICAC strategy. In the transition diagram, we assume that the original network capacity is C, and is the maximum data traffic ratio in the network, i.e. the data traffic cannot occupy more than αC capacity. We define λ_1 and λ_2 are the Poisson arrival rate of voice traffic and data traffic, and μ_1 and μ_2 are the exponential dispatch rate of voice traffic and data traffic.

In the system which do not use ATICAC strategy shown in Figure 4, let i denote the number of data traffic and j the number of voice traffic. The the state space of the two-dimensional Markov chain is given as following

$$S = \{(i, j) \mid 0 \le i \le \alpha C, \quad 0 \le j \le C, i + j \le C\}. \tag{17}$$

Let $p_{i,j}$ denote the steady-state probability when there are i data traffic and j voice traffic. From the the balance equation, we get

$$\begin{cases} p_{i,j-1}\lambda_1 = p_{i,j}\mu_1 \\ p_{i-1,j}\lambda_2 = p_{i,j}\mu_2 \end{cases}. \tag{18}$$

Figure 3 Admission control implementation

Algorithm: Admission Control Scheme

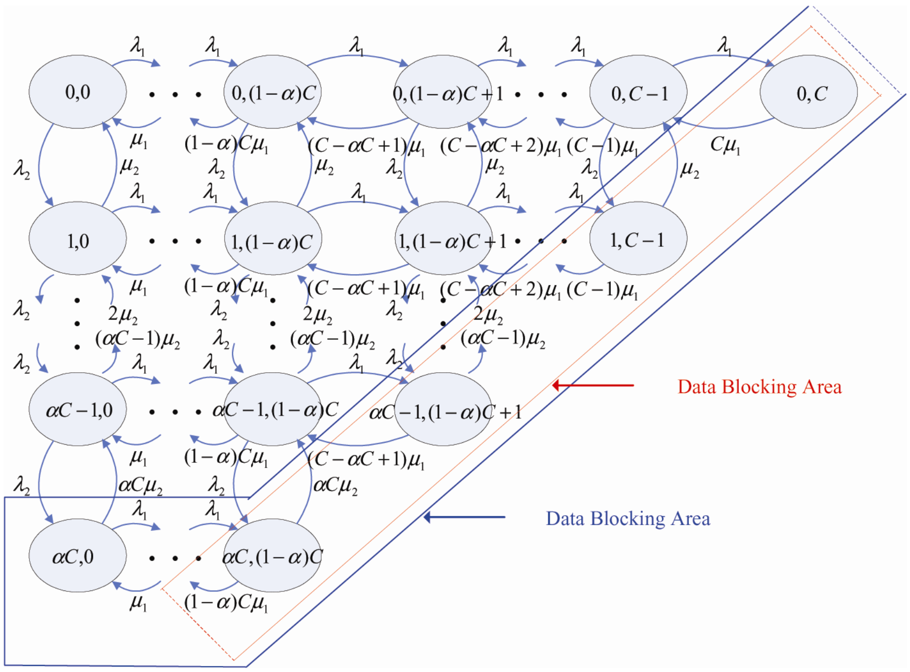
Input: The action of coming request **"Request_Action'**; Type of request traffic **"Traffic_Type'**; Transmitting interval of existing calls **"Interval'**; Existing data traffic capacity **"Data_Capacity'**.

Output: Whether to admit the coming traffic or not;
Whether to enhance the quality of existing calls.

Method:

```
IF Request_Action == Coming
Calculate collision probability p_c through p_b
    IF Traffic_Type == Voice
        IF p_c < p'_c
            Admit this voice call
        ELSE
            IF Interval ≥ 50ms
                Reject this voice call
            ELSE
                Interval ← interval + 10ms
                Admit this voice call
            ENDIF
        ENDIF
    ENDIF
    IF Traffic_Type == Data
        IF p_c < p'_c and Data_Capacity< αC
            Admit this data traffic and
            Data_Capacity ← Data_Capacity + 1
        ELSE
            Reject this data traffic
    ENDIF
ELSE
    IF Traffic_Type == Data
        Data_Capacity ← Data_Capacity − 1
    ENDIF
    Interval ← interval − 10ms
    Calculate collision probability p_c through p_b
    IF p_c ≥ p'_c
        Interval ← interval + 10ms
    ENDIF
ENDIF
```

Figure 4 Transition diagram for without deploying Adaptive Transmission Interval Call Admission Control strategy (see online version for colours)



The red line denotes the set of blocking states for data traffic; The blue line denotes the sets of blocking states for voice traffic. λ_1 (resp. μ_1) and λ_2 (resp. μ_2) are the call arrival (resp. service) rates of voice and traffic, respectively.

We can obtain

$$p_{i,j} = p_{0,0}\rho_2^j \rho_1^j \frac{1}{i!\,j!} \tag{19}$$

where $\rho_1 = \dfrac{\lambda_1}{\mu_1}, \rho_2 = \dfrac{\lambda_2}{\mu_2}$. From the normalisation equation, we obtain

$$p_{0,0} = \frac{1}{\sum_{i=0}^{ac} \rho_2^i \frac{1}{i!} \sum_{j=0}^{C-i} \rho_1^j \frac{1}{j!}} \tag{20}$$

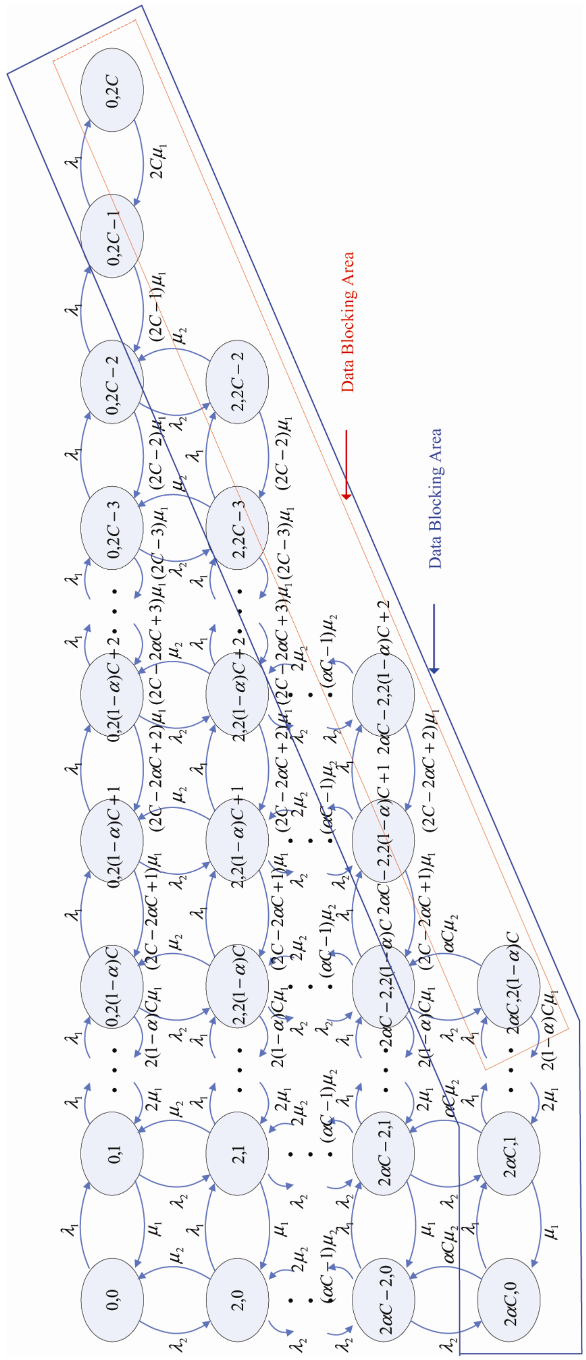
From the above diagram, the states which are in the scope of red line are the states which the voice traffic should be blocked and the sates which are in the scope of blue line are the states which the data traffic should be blocked.

Let S_{vb} and S_{db} denote the subspaces of S with the voice traffic blocked and the data traffic blocked, respectively. Then we obtain

$$S_{\text{vb}} = \{(i,j) \mid 0 \le i \le \alpha C, \quad 0 \le j \le C, i+j = C\} \tag{21}$$

$$S_{\text{db}} = \{(i,j) \mid 0 \le i \le \alpha C, \quad 0 \le j \le C, i+j = C \text{ or } i = \alpha C\}. \tag{22}$$

Figure 5 Transition diagram for Adaptive Transmission Interval Call Admission Control strategy (see online version for colours)



The red line denotes the set of blocking states for data traffic; The blue line denotes the sets of blocking states for voice traffic. λ_1 (resp. μ_1) and λ_2 (resp. μ_2) are the call arrival (resp. service) rates of voice and traffic, respectively.

We can also obtain the formulas for voice traffic blocking probability p_{vb} and data traffic blocking probability p_{db} as follows

$$p_{\mathrm{vb}} = \sum_{i=0}^{\alpha C} \sum_{j=C-i} p_{i,j} = \frac{\sum_{i=0}^{\alpha C} \rho_2^i \rho_1^{(C-i)} \frac{1}{i!} \cdot \frac{1}{(C-i)!}}{\sum_{i=0}^{\alpha C} \rho_2^i \frac{1}{i!} \sum_{j=0}^{C-i} \rho_1^j \frac{1}{j!}} \tag{23}$$

$$\begin{aligned} p_{\mathrm{db}} &= \sum_{i=0}^{\alpha C} \sum_{j=C-i} p_{i,j} + \sum_{i=\alpha C} \sum_{j=0}^{C-\alpha C-1} p_{i,j} \\ &= \frac{\sum_{i=0}^{\alpha C} \rho_2^i \rho_1^{(C-i)} \frac{1}{i!} \cdot \frac{1}{(C-i)!} + \sum_{j=0}^{C-\alpha C-1} \rho_2^{\alpha C} \rho_1^j \frac{1}{(\alpha C)!} \cdot \frac{1}{j!}}{\sum_{i=0}^{\alpha C} \rho_2^i \frac{1}{i!} \sum_{j=0}^{C-i} \rho_1^i \frac{1}{j!}} \end{aligned} \tag{24}$$

From Table 1, we can show that the system capacity for the voice traffic enlarges to $2C$ using ATICAC strategy, and the capacity for the data traffic remains the same as before. In order to calculate the respective blocking probabilities in the system where the ATICAC strategy is applied, we get the two-dimensional Markov chain as shown in the Figure 5. Let i' denote the number of data traffic and j' the number of voice traffic. Then, the state space of the two-dimensional Markov chain is given as follows

$$S' = \{(2i', j') \mid 0 \le i' \le \alpha C, 0 \le j' \le 2C, 2i' + j' \le 2C\}. \tag{25}$$

Let $p'_{2i',j'}$ denote the steady-state probability when there are i' data traffic and j' voice traffic. Then, the balance equation becomes

$$\begin{cases} p_{2i',j'-1}\lambda_1 = p_{2i',j'}\mu_1 \\ p_{2i'-2,j'}\lambda_2 = p_{2i',j'}\mu_2 \end{cases}. \tag{26}$$

We can obtain

$$p_{2i',j'} = p_{0,0}\rho_2^{i'}\rho_1^{j'} \frac{1}{i'!\,j'!}. \tag{27}$$

From the normalisation equation, we obtain

$$p_{0,0} = \frac{1}{\sum_{i'=0}^{\alpha C} \rho_2^{i'} \frac{1}{(i')!} \sum_{j'=0}^{2C-2i'} \rho_1^{j'} \frac{1}{j'!}}. \tag{28}$$

From the above diagrams, we understand that the states which are in the scope of red line are the states in which the voice traffic should be blocked and the sates which are in the scope of blue line are the states in which the data traffic should be blocked.

Let S'_{vb} and S'_{db} denote the subspaces of S' with the voice traffic blocked and the data traffic blocked, respectively. Then, we obtain

$$S'_{\text{vb}} = \{(i', j') \mid 0 \le i' \le \alpha C, \quad 0 \le j' \le 2C, 2i' + j' = 2C\} \tag{29}$$

$$S'_{\text{db}} = \{(i', j') \mid 0 \le i' \le \alpha C, 0 \le j' \le 2C, \quad 2i' + j' = 2C \text{ or } 2i' + j' = 2C - 1 \text{ or } i' = \alpha C\}. \tag{30}$$

From this, we obtain the formulas for voice traffic blocking probability p'_{vb} and data traffic blocking probability p_{db} using ATICAC strategy as follows, or

$$p_{\text{vb}} = \sum_{i'=0}^{\alpha C} \sum_{j'=2C-2i'} p_{2i',j'} = \frac{\sum_{i'=0}^{\alpha C} \rho_2^{i'} \rho_1^{(2C-2i')} \frac{1}{i!} \cdot \frac{1}{(2C-2i)!}}{\sum_{i'=0}^{\alpha C} \rho_2^{i'} \frac{1}{i'!} \sum_{j'=0}^{2C-2i'} \rho_1^{j'} \frac{1}{j'!}} \tag{31}$$

$$\begin{aligned} p_{\text{db}} &= \sum_{i'=0}^{\alpha C} \sum_{j'=2C-2i'-1'}^{2C-2i'} p_{2i',j'} + \sum_{i'=\alpha C} \sum_{j'=0}^{2C-2\alpha C-2} p_{2i',j'} \\ &= \frac{\sum_{i'=0}^{\alpha C} \rho_2^{i} \frac{1}{i'!} \left(\rho_1^{2C-2i'} \frac{1}{(2C-2i')!} + \rho_1^{2C-2i'-1} \frac{1}{(2C-2i'-1)!} \right)}{\sum_{i'=0}^{\alpha C} \rho_2^{i'} \frac{1}{i'!} \sum_{j'=0}^{2C-2i'} \rho_1^{j'} \frac{1}{j'!}} \\ &\quad + \frac{\sum_{j'=0}^{2c-2\alpha c-2} \rho_1^{j'} \frac{1}{j'!} \rho_2^{\alpha C} \frac{1}{(\alpha c)!}}{\sum_{i=0}^{\alpha C} \rho_2^{i'} \frac{1}{i'!} \sum_{j'=0}^{2C-2i'} \rho_1^{j'} \frac{1}{j'!}}. \end{aligned} \tag{32}$$

6 Simulations and numerical results

In this section, we will evaluate the proposed ATICAC through extensive NS-2 Simulator (2007) simulations. We will verify the analytical results for VoIP in WLANs given in Section 3. Furthermore, we will show that our call admission control strategy with adaptive transmission interval given in Section 4 can improve the performance of VoIP over IEEE 802.11 WLANs in terms of more admitted calls, smaller delay and almost zero packet loss rate.

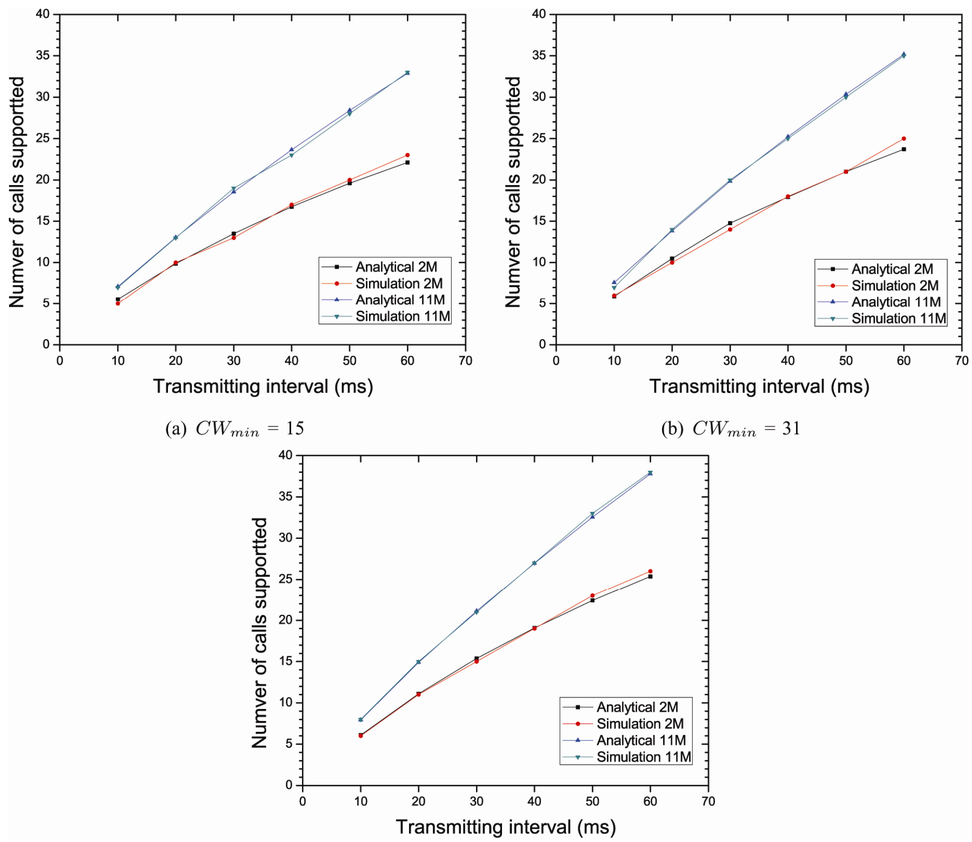
6.1 Simulation environments

In our simulation study, the radio propagation model is a two-ray ground model with a wireless LAN using IEEE 802.11 MAC protocol with direct sequence spread spectrum. Other defaults parameters are summarised in Table 1. We use different numbers of mobile stations in a rectangular grid with its dimension being 150 × 150 m^2 to simulate the wireless LAN. We use CBR (Constant Bit Rate) traffic to simulate half duplex VoIP connections.

6.2 Model validation

Figure 6 shows the maximum number of calls vs. transmission interval, using different $CW_{\min}$ and data rate of the network, to compare our analytical model with simulation results. We obtain the analytical results using the analytical model for VoIP in 802.11 WLAN in Equations (2), (10), (11) and (13) to be compared with ns-2 simulation results. We can see that the analytical results closely match the simulations for network data rate of 2 or 11 Mbps, with varieties of $CW_{\min} = 15$, 31 and 63. The maximum number of call supported increases with the increase of transmitting interval. We can also see that $CW_{\min}$ has a little effect on the maximum number of calls supported in the network because a larger $CW_{\min}$ provides more backoff timer selections. Therefore, it can decrease the average collision probability within a certain range. In our analytical model and simulation results, increasing $CW_{\min}$ from 15 to 63 can allow one more call to join the network at most in some situations. This suggests that our analytical model is robust against $CW_{\min}$.

Figure 6 Maximum number of calls supported with transmission interval increasing (see online version for colours)



(a) $CW_{min} = 15$

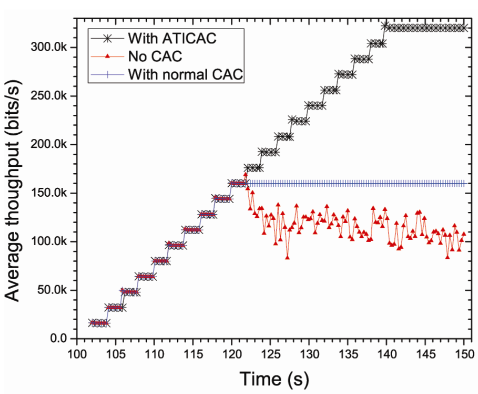
(b) $CW_{min} = 31$

(c) $CW_{min} = 63$

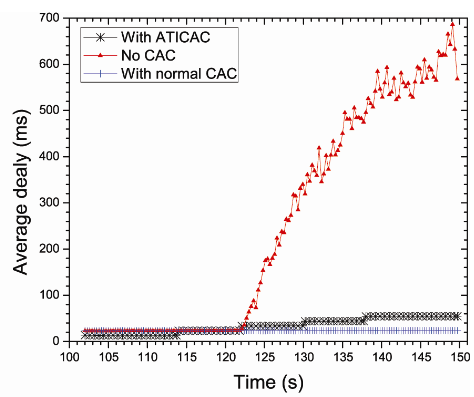
6.3 Performance evaluation through simulations

To evaluate the performance of ATICAC, we consider a scenario that VoIP calls join the network one by one every two seconds from 102 sec. All calls in the simulation use fixed 20 ms transmission interval which is wildly used for VoIP calls (International Telecommunication Union Telecommunication Standardisation Sector, 2007), except for ATICAC. In Figure 7, the line with legend star represents the network performance when there is no call admission control. After the 11th call which joins the network at the time 122 sec, the network falls into saturate mode. The average throughput decreases slightly and the average delay and average loss rate increase dramatically in Figure 7(b) and (c). The line with legend triangle represents the network performance using normal call admission control. When the number of existing calls in the network attains the maximum value, i.e. 10 calls, the BS will reject new arriving calls until one existing call finishes. The normal call admission control strategy can guarantee the quality of existing calls, while the number of ongoing calls is still small and the capacity of WLAN for VoIP is not fully explored. Compared with normal CAC strategy, our ATICAC strategy (with legend cross) allows twice simultaneous calls as that of the normal CAC strategy, i.e.

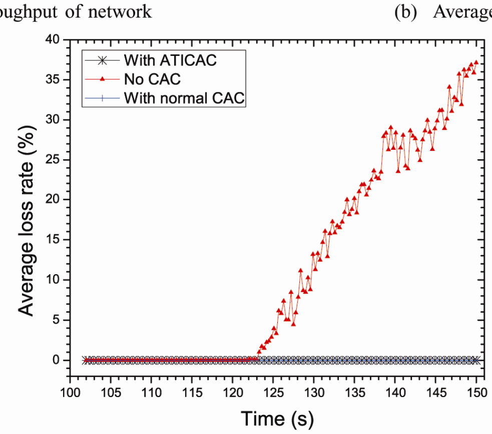
Figure 7 Call quality statistics with the number of calls increase (see online version for colours)



(a) Average throughput of network



(b) Average delay of calls

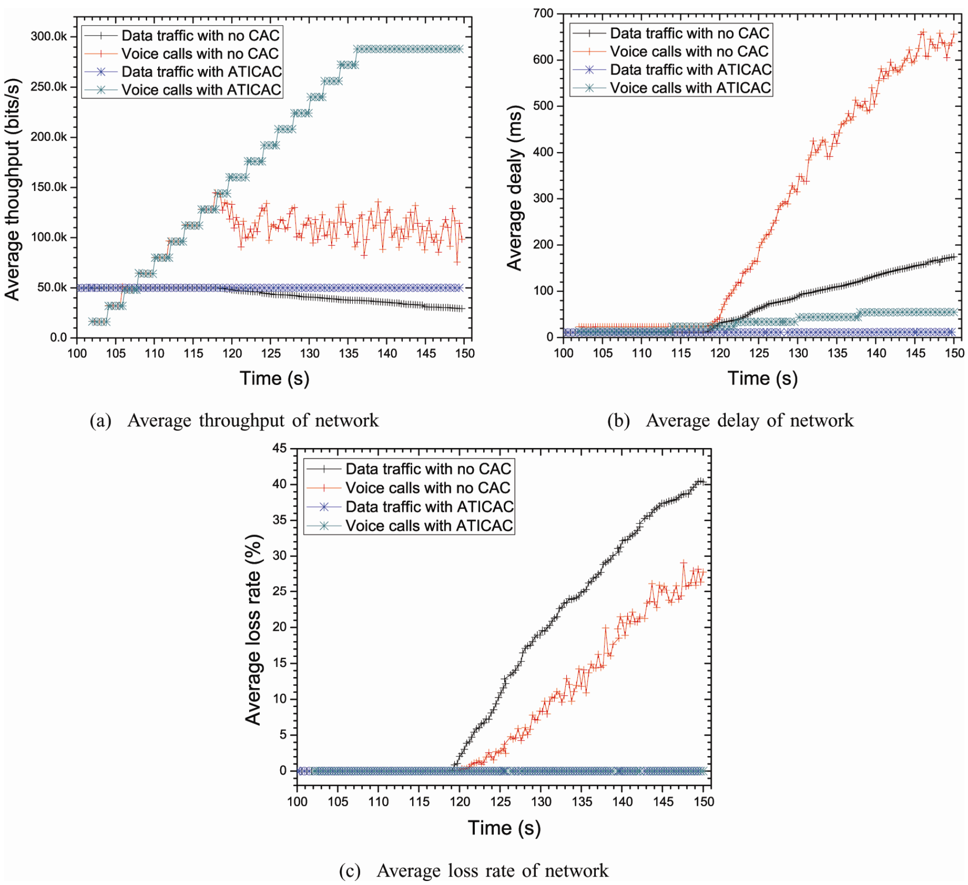


(c) Average loss rate of calls

100% improvement, at the expense of 30 ms delay increase. We note that the increase of 30 ms for delay incurs 0.72 decrease of **R**-score, which is still in the tolerable range of VoIP call as we have discussed in Section 3. On the other hand, if there are small number of calls in the network, for example, less than six calls in the network, the average delay of the ATICAC strategy is smaller than normal CAC strategy due to smaller transmission interval which adaptively decreases.

We also examine ATICAC strategy when there exists background data traffic in the network as shown in Figure 8. The data traffic starts from 100 to 150 sec with a rate of 50 kbps. The VoIP calls join the network one by one every two seconds from 102 sec. We can observe from Figure 8 that the VoIP calls and data traffic using ATICAC strategy are in good quality all the time if compared with the situation that no CAC strategy is used. Compared with the above simulations which do not have any data traffic, the maximum number of calls allowed in the network decreases from 20 to 18 due to the data traffic, but the QoS of calls remains the same.

Figure 8 Network statistics with the number of calls increase (see online version for colours)



(a) Average throughput of network

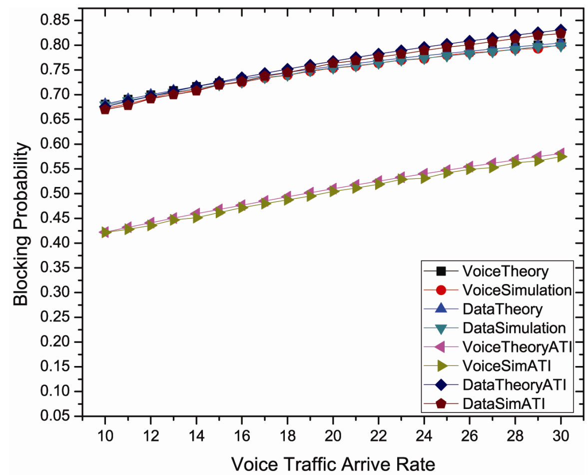
(b) Average delay of network

(c) Average loss rate of network

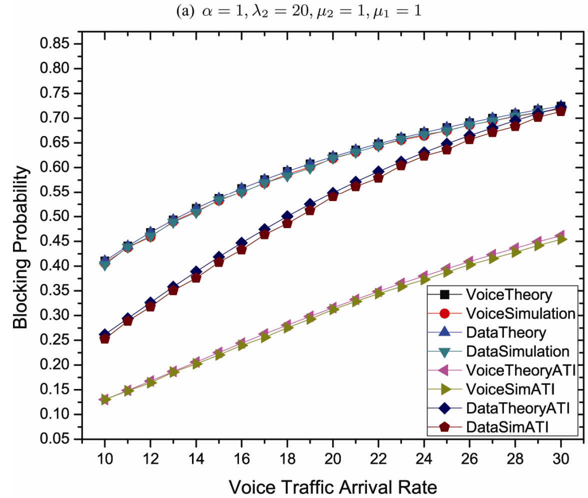
6.4 Performance validation through numerical results

To evaluate the performance of ATICAC, we consider a scenario that VoIP calls and data traffic arrival rate are Poison distributed with parameters λ_1 and λ_2. The exponential dispatch time for VoIP calls and data traffic are μ_1 and μ_2. From our analytical study and simulation, we know that the system capacity for VoIP calls and data are 10 and 10α with parameters shown in Table 1. Figures 9 and 10 show the voice and data traffic blocking probability with voice traffic arrival rate increasing from 10 to 30 derived from Equations (23), (24), (31) and (31) as well as from event driven simulations.

Figure 9 Blocking probability vs. voice traffic arriving rate (see online version for colours)

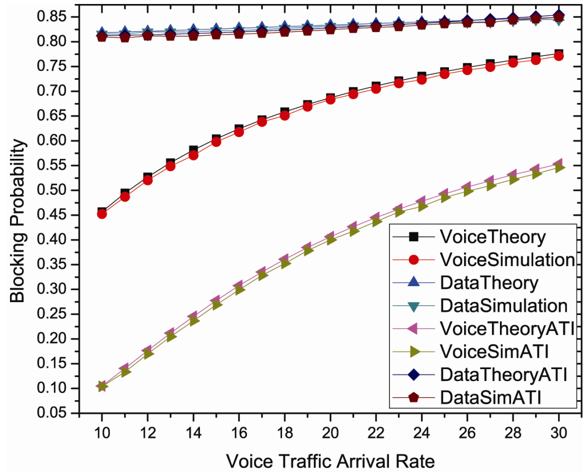


(a) $\alpha = 1, \lambda_2 = 20, \mu_2 = 1, \mu_1 = 1$

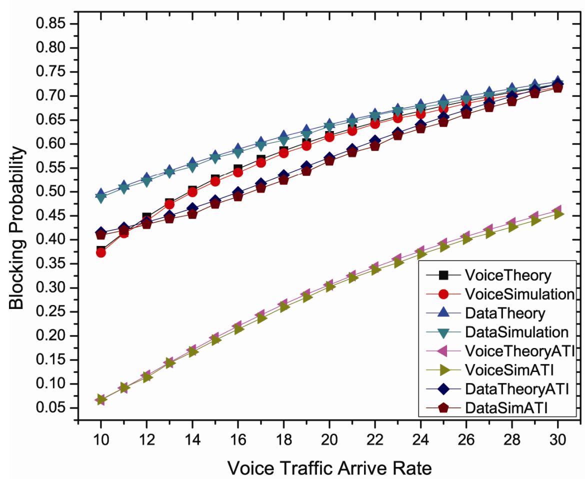


(b) $\alpha = 1, \lambda_2 = 10, \mu_2 = 2, \mu_1 = 1$

Figure 10 Blocking probability vs. voice traffic arriving rate (see online version for colours)



(a) $\alpha = 0.4, \lambda_2 = 20, \mu_2 = 1, \mu_1 = 1$

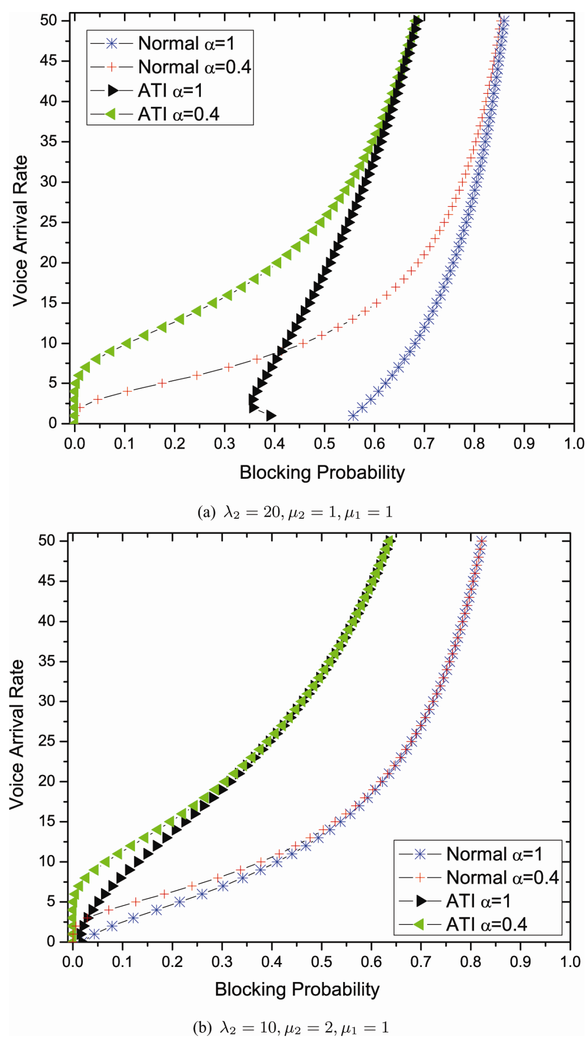


(b) $\alpha = 0.4, \lambda_2 = 10, \mu_2 = 2, \mu_1 = 1$

We first show the results when α equals one in ATICAC strategy which means that there is no restriction with data traffic in the system. From Figure 7a and b, we can obtain that the ATICAC strategy can reduce 50% voice traffic blocking probability with different traffic arrival rates and dispatch times. Especially if the traffic in the system has a low arrival rate and short dispatch time, ATICAC will also reduce 50% data traffic blocking probability. If we restrict data traffic to 40% of system capacity, i.e. α equals to 0.4 in ATICAC strategy as shown in Figure 7 (c) and (d), we can obtain that using ATICAC strategy the voice blocking probability can be reduced 80% if compared with the case that no ATICAC strategy is used, and 50% if compared with the strategy with a restrict data traffic up to 40% system capacity. Moreover, the voice blocking probability will be reduced more dramatically using ATICAC strategy if the traffic load in the system is very high. To better evaluate the performance of ATICAC strategy, we show the maximum voice traffic arrival rate a system can support with certain voice traffic blocking probability in Figures 9 and 10 using system parameter shown in Table 1 and different data traffic arrival rates. We can obtain that using ATICAC strategy, the system can accommodate at least two times of traffic with a fixed blocking probability, which means we can allow more voice calls in the network.

As a final remark, changing the parameter α in ATICAC strategy will also have some influence to the performance of the network. We can obtain from Figure 7 that a smaller α will increase the blocking probability of data traffic and decrease the blocking probability of voice traffic, no matter whether the traffic load in the network is heavy or not. Therefore, we should choose a smaller α if the voice traffic blocking probability is larger than our expectation. We can also see that the blocking probability is more sensitive with α if the traffic load in the network is heavy. Therefore, we should adjust α with a smaller step if the network traffic load is heavy; otherwise we should change α with a larger step in a light traffic load in order to adjust the blocking probability more quickly (Figure 11).

Figure 11 Maximum traffic arriving rate vs. blocking probability (see online version for colours)



(a) $\lambda_2 = 20, \mu_2 = 1, \mu_1 = 1$

(b) $\lambda_2 = 10, \mu_2 = 2, \mu_1 = 1$

7 Conclusions

In this article, we study how IEEE 802.11 WLANs can support more VoIP calls, and we provide a simple analytical model, which is able to calculate the maximum number of VoIP calls supported in IEEE 802.11 WLANs to accurately match with simulation results. Using the proposed model, we find that the high collision probability takes the main responsibility to the decrease of VoIP calls if too many calls join the network. Thus, to support a given Qos of VoIP calls, the IEEE 802.11 WLANs should not work in the saturate mode which has a higher collision probability.

We have proposed a new call admission control strategy, ATICAC, in which the BS controls the average collision probability p_c of the network to prevent the network from saturation. BS will adaptively change the transmission interval of active stations when a call arrives or leaves. ATICAC can ensure the QoS of VoIP calls in 802.11 WLANs, and at the same time can provide more VoIP calls for a given WLAN. We have also evaluated the performance of ATICAC through extensive ns-2 simulations. The results indicate that ATICAC can support 50–100% more VoIP calls according to different scenarios, which matches the analytical results very well. As to the same number of calls, ATICAC provides up to 50% less delay than the normal CAC strategy. Therefore, the proposed ATICAC improves VoIP over WLANs by carefully leveraging the tradeoff of voice delay and number of calls through adaptive transmission interval.

Acknowledgements

This work was supported in part by NSFC (No. 60702046), China Ministry of Education (No. 20070248095), Shanghai Jiaotong University Young Faculty Funding, Shanghai Jiaotong University Pre-Research Funding; Qualcom China Research Grant and Taiwan National Science Council (NSC96-2221-E-110-035 and NSC96-2221-E-110-050).

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