

Performance Improvement of Voice over Multihop 802.11 Networks

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Abstract—Extensive studies on supporting voice traffic over wireless 802.11 networks have been carried out in the literature. Most of them were focused only on one hop infrastructure mode. This paper addresses the issue of voice over multihop wireless networks. Through simulation considering a simple topology, we identify the voice traffic bottleneck in 802.11 MAC and thus propose a Burst Queue (BQ) scheme to cooperate with 802.11b and 802.11e MAC protocols. Extensive simulation results under variety of circumstances show that BQ scheme can support 50% ~ 100% more voice calls and obtain 20% ~ 50% lower delay at the cost of a little higher loss ratio which is tolerable to voice flows. Even for the grid and random topology, our BQ scheme can provide smaller packet loss ratio, and shorter end-to-end delay.

I. INTRODUCTION

With the rapid deployment of IEEE 802.11 networks, extensive research efforts have been carried out on voice applications over wireless networks. Currently, most of existing studies deal with infrastructure mode or one-hop circumstance, and only a few works have investigated the voice over *multihop* ad hoc networks [1], [2]. Since multihop networks are more flexible and have a wide range of applications like wireless sensor network [15], [16], it is important to explore the potential of voice applications in these circumstance.

It is well known that voice traffic has two significant features different from the none-voice flows: (i) The first feature is the small payload size. In general, a packet is composed of header and payload. The packet header is used to record the control information on how to transmit the data, and the payload is the real information that we want to deliver. Since the overhead for the packet header is almost fixed for each packet, it is common to use a large payload size to increase the network efficiency. However, for the nature of voice traffic, the payload size is always small (even smaller than the header size). This leads to the low channel utilization of the networks; (ii) The second difference depends on the nature of human being's hearing. Voice traffic should meet the delay requirement (usually less than 200 ms), while it is able to tolerant up to 4% packet loss ratio.

In this paper, we investigate the voice traffic over multihop 802.11 networks. Through simulation studies, we find that one of the major bottlenecks of the networks is the IEEE 802.11 MAC layer. The bottleneck node suffers sever MAC layer queueing overflow, which incurs lots of packet loss, and

hence degrades the performance of voice traffics. With this observations, we propose a burst queue (BQ) scheme to reduce the packet loss and improve the performance. The idea is that the bottleneck nodes pack several packets in their queue into a large packets and broadcast them all in once to increase the network efficiency. Each node monitors its queue length and decides whether to use BQ or not. We simulate various traffic patterns over BQ scheme and our simulation results show the superiority over the original IEEE 802.11 MAC standard. The novelty of this paper is that we change the legacy burst method from one to one burst to one to many. The idea is to let the sender send once and let the receiver unpack the packet and pick up the information destined for it.

We organize the rest of the paper as follows. We present the background and related previous work in Section II, and describe the problem for multihop 802.11 networks in Section III. Then, we propose the BQ scheme in Section IV. We provide the extensive simulation results under various traffic patterns in Section V, and we conclude our work in Section VI.

II. BACKGROUND AND RELATED WORK

The ultimate objective of VoIP is to deliver high-quality voice service, which is comparable to what is provided in traditional circuit-switching networks. When considering the problem of transmitting VoIP traffic over wireless networks, numerous challenges are encountered. Due to the deficiency in the wireless media access methods, the delivery of VoIP often leads to unpredictable delay and packet-loss [1] performances.

In recent years, extensive research and development efforts have been conducted on IEEE 802.11 networks [2], [3], [4]. In [4], Sachin Garg et al, present experimental studies on the throughput of IEEE 802.11b wireless networks and point out that the inherent channel inefficiency limits the maximum number of voice flows. Their experiments revealed that the aggregated bandwidth of the networks is diminished by ongoing VoIP connections. Bianchi et al proposed a Markov chain model for the binary exponential backoff procedure [5]. Based on the saturated throughput derived from the model, Foh et al. obtain the mean packet delay [6]. In [7], the authors use a p -persistent backoff strategy to approximate the original back off in the protocol.

Since 802.11b can only support best effort traffic, 802.11e is designed to make up for the deficiency. The creation of 802.11e EDCA is due to extensive research that aimed to support prioritized service over 802.11 DCF [8], [9]. There are even some works [10], [11] that try to enhance the performance of 802.11e. However, most of these works are focused on the priority of different traffics which means if we just transmit voice flows over the networks, there would be no difference between DCF and EDCA. So it is necessary to study the networks that transmit voice packets only. The idea is that if a scheme can support more VoIP flows in a voice-only environment, its performance should be better in none-voice-only (with EDCA) conditions.

For voice flows, delays longer than $100ms$ require echo cancellation. Long delays would lead to cross-talk. The regular retransmission mechanism also results in excessive end to end delays that are not desirable. However, voice can afford a small amount of packet loss since the ear-brain is less sensitive to short dropouts in received speech [12]. According to this characteristic of voice, we develop the BQ scheme through which, we improve the delay and throughput dramatically while the loss ratio is still kept within an acceptable level.

III. PROBLEM DESCRIPTION

The most significant difference between VoIP traffic and other flows is the payload size. The payload size of VoIP is from 33 bytes (GSM) to 160 bytes (G.711), while TCP payload size is generally 1460 bytes. Since the transmission overhead is fixed for each packet, lower payload size results in lower channel utilization. Consider a VoIP packet with 33-byte payload, the transmission time for it at 11 Mbps is $33 \times 8 \div 11 = 24\mu s$. The transmission time for the 40-byte IP/UDP/RTP header is $40 \times 8 \div 11 = 29\mu s$. However, the 802.11 MAC and PHY layers have additional overhead of more than $800\mu s$, attributed to the physical preamble, MAC header, MAC backoff time and MAC ACK [13]. Thus, the overall channel efficiency is less than 3%. Note that this calculation already eliminates the CTS/RTS hand shake overhead of the 802.11 MAC of Unidirectional VoIP Flows

Another problem incurred by multihop networks is that topology may affect the performance of VoIP calls. When we simulate 10 VoIP traffics in a flat area with random positioned nodes, the total throughput differs for each random topology. In some conditions, the throughput degrades dramatically while in others the performance is much better. Then, we investigate the poor performance conditions and find that it is the busiest node in the whole network that affects the performance most.

Thus, if we increase the number of the calls, the send buffer (queue) of node A, B and C will first overflow which results in an unacceptable packet loss ratio. This is due to the extreme low channel utilization and rather high packet arrival rate. We name node A, B and C as the bottleneck of the whole network in Figure 1. If we want to provide more VoIP calls in multihop condition, we must first solve the problems of the bottleneck.

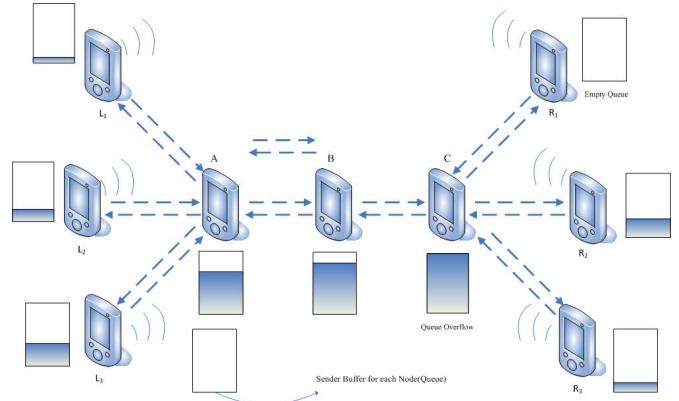


Fig. 1. Bottleneck of a common network for VoIP

In the rest of this paper, we would take node A, B and C as our basic bottleneck model.

IV. BURST QUEUE SCHEME

As in the bottleneck model, busy nodes have a long backlog, each packet in the queue will be sent in a 3% efficient channel which finally results in the queue overflow. To improve the channel utilization, we proposed a Burst Queue scheme to increase the channel efficiency as well as reduce the delay. Since the overhead of MAC and PHY are too high, we try to combine several transmissions into one burst broadcast to save this overhead. That is to say, when the backlog is too long, the MAC would be notified. This notification would be explained as a warning of packet overflow or undesirable high delay. Then the MAC would pack the packets in the queue to make a big packet whose size would not exceed 1200 bytes. Queue length represents the business of the node. If the queue length is kept at a low level, the MAC performs default operation. If the queue length exceeds the WATER_MARK (we set it to 10), the queue notifies the MAC and the BQ scheme starts up. The BQ broadcast packet contains just one MAC header, but each packet is packed with its destination MAC address ahead. The receiving MAC uses this information to decide whether to deliver the packet to the up layer or just discard it. This burst is one to many which is different from the legacy one to one burst.

If we pack n packets each time, we save about $40n$ bytes extra MAC and PHY header, $n - 1$ backoff time and n MAC ACK time at the cost of lower transmission rate for broadcast and a higher collision probability. Our simulation results show that the cost is worthy. With BQ scheme, the bottleneck model could support more VoIP calls and obtain a lower delay without queue overflow. For non-bottle-neck nodes which have a short queue, burst could never happen. They preserve the reliability of the 802.11 retransmission and ack mechanism. Our idea is that if the node is not busy, we care more about reliability; otherwise, we pack several packets in the queue and make a burst broadcast to increase the channel reuse. This is a tradeoff between contention and collision to

prevent the queue from overflow and acquire a lower delay to satisfy the VoIP traffic.

Algorithm: Burst Queue Scheme

Input: The sender buffer queue length **QueLen**; the maximum packet length **BURST_SIZE**; the queue monitor **WATER_MARK**.

Output: Judge whether to use BQ scheme.

Method:

1. if **MAC_STATE == IDLE**
2. if **QueLen < WATER_MARK**
3. **Deliver** packet 1 to Mac layer
4. Mac add common Mac header
5. else
6. Pack K packets in the queue
7. // $\text{WATER_MARK} \leq K \leq \text{QueLen}$
8. // $\sum_{i=1}^K \text{sizeof(packet } i) \leq \text{BURST_SIZE}$
9. **Deliver** the Packed packet to Mac
10. Mac add a broadcast header
11. endif
12. endif

Fig. 2. The Burst Queue Scheme Implementation.

V. SIMULATION AND RESULTS

A. Simulation setup

We simulate the bottleneck nodes and develop a testbed with nodes in a chain topology, each node is 200m away and have a 250m transmit range and 550m sensing range. To simulate the bottleneck condition, we send multiple aggregated flows from each end of the node, thus the middle node would become the bottleneck node. We use CBR traffic as our VoIP flow, the payload size is 33 bytes, interval is 20ms, and data rate is 13.2Kbps as specified in GSM. We run a TCP flow of Reno version over the test model whose packet size is set to 1024 bytes. All simulations last 100 seconds in NS-2 [14].

B. BQ over 802.11b

We first simulate unidirectional condition. We transmit N aggregated VoIP flows from node A to node C (as shown in Figure 1).

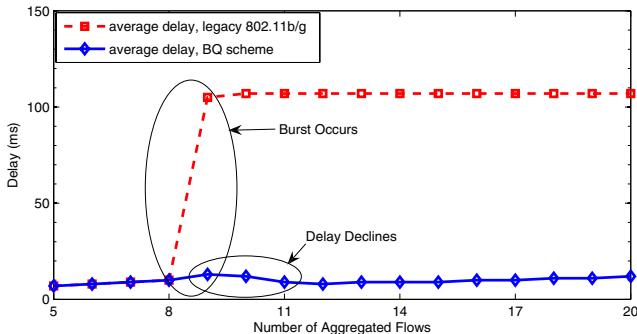


Fig. 3. Packet Delay of Unidirectional VoIP Flows over Bottleneck Model.

As shown in Figures 3 and 4, when the number of flows is 5, broadcast does not happen which means that the queue of the

three node never exceeds 10. In these unsaturated networks, our scheme acts the same as legacy 802.11b protocol. Then we increase the number of the flows, when $N = 6$ and 7, burst broadcast still does not occur. When N reaches 8, burst broadcast happens once under BQ scheme. We can see from the Figure 3 that average delay of BQ is much lower than that of legacy 802.11b. When we increase one more voice flow, the three-node network becomes the bottleneck model which is exactly we expect. Due to the queue overflow, legacy 802.11b network has a loss ratio of 3.3% which almost exceeds the voice transmission requirement. While in BQ scheme, with 2043 times burst broadcast within the simulation period of 100 seconds, the queues of three nodes never overflow. The 0.06% loss ratio is due to the collision.

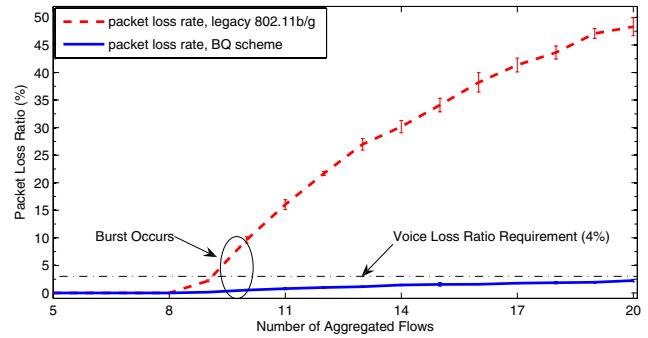


Fig. 4. Packet Loss Ratio with 95% Confidence Interval of Unidirectional VoIP Flows over Bottleneck Model.

The average delay of BQ in Figure 3 is highly improved. Then we increase N to 20, which is rather large, we can see that with the queue overflow, legacy 802.11b suffers from a high packet loss rate up to 46.5%, while BQ just reaches the loss ratio similar to $N = 9$ situation of legacy 802.11b. We can see that in Figure 4, BQ still keeps a low average delay which fully satisfies the voice transmission requirement. Note the average delay of legacy 802.11b remains almost constant (106ms) as shown in Figure 4 from 9 to 20 flows. This is due to the fact that when the network is saturate, the packet transmission rate reaches its up-limit, and the extra packets are dropped due to the queue overflow. Thus, the average duration of each packet in the queue is the same. For BQ, we observe that the average delay for 11 voice flows is even lower than that of 9 and 10 flows. This is because the busier the network is, the more the BQ scheme takes place. For 11-flow condition, the queue length of each node almost always exceeds the **WATER_MARK** while for 9 and 10 flows conditions, BQ occurs not as frequently as 11-flow condition. Though BQ decreases the probability of queue overflow, if we keep increasing the number of flows, the packet loss ratio of BQ will also increase and the queue of our scheme will overflow eventually (e.g. 50 flows).

Now we examine the simulation results for bidirectional flows. There are M voice flows transmitting from each end to the other end (A to C and C to A, as shown in Figure 1).

Hence, the total number of flows would be $2M$.

Like the unidirectional condition, Table I shows the superiority of BQ scheme as well. When $M = 4$, the network is unsaturated and burst does not happen. When M reaches 5, burst occurs 156 times and BQ reduces the average delay by 81.3%. Legacy 802.11b can support only 5 pure voice pairs, and its delay is rather high, while BQ over 802.11b could support 7 pure voice pairs and still achieve a low average delay at 21ms. The nodes A, B and C become the bottleneck when M reaches 6. The queue overflow results in the high loss ratio of 802.11b, we also change the queue length to 500, but it becomes not helpful and the delay exceeds 400ms which is apparently not acceptable. We note that the improvement in delay leaves sufficient margin for further transmission, since the nodes in both end (node A and node C) may not be the terminal of the whole transmission. This extra delay could be used for the transmission over Internet etc.

Finally, we add a background TCP flow to the bidirectional condition. The TCP flow is transmitted from node A to node C. The results are shown in Table I. Without the support of priority service, the performance of the VoIP degrades dramatically. The aggressiveness of TCP leads to more contentions. BQ scheme achieves the tradeoff between contention and collision. With a little more collision, BQ reduces contention and delay of the flow.

We observe that legacy 802.11b MAC has a lower packet loss ratio, but the delay is too high to support voice. If we take a look at the throughput achieved by TCP and the average delay. We can conclude that BQ is superior. It is the TCP that affect the performance of voice. Aggressive TCP flow results in more contentions and thus results in more broadcast failure. (Some TCP packet packed with voice packets, and broadcast out).

TABLE I
LOSS RATIO, DELAY, BURST BROADCAST TIMES COMPARISON

M pairs voice		4	5	6	7
legacy 802.11b	loss ratio	0%	1.5%	24.1%	34.8%
	ave. delay (ms)	-	96	268	275
	max delay (ms)	-	156	353	364
BQ over 802.11b	loss ratio	0%	0.1%	0.06%	3.0%
	ave. delay (ms)	-	18	23	21
	max delay (ms)	-	71	92	92
	burst times	0	153	2615	4717
M pairs voice with 1 TCP		3	4		
legacy 802.11b	loss ratio	1.8%		3.6%	
	ave. delay (ms)	117		111	
	max delay (ms)	303		362	
	TCP thp. (Kbps)	265.25		136.99	
BQ over 802.11b	loss ratio	3.1%		3.9%	
	ave. delay (ms)	45		39	
	max delay (ms)	182		133	
	burst times	3761		5466	
	TCP thp. (Kbps)	400.91		365.45	

C. BQ over 802.11e

With a TCP flow, 802.11b becomes not suitable for voice services, so we implemented BQ scheme in 802.11e. We implement it in the queue of AC0 which is designed to transmit voices. We assign VoIP to AC0, and TCP to AC3 and run the simulation again.

TABLE II
LOSS RATIO, DELAY, BURST BROADCAST TIMES COMPARISON

M pairs flow with 1 TCP		4	5	6
legacy 802.11e	loss ratio	0%	0.6%	20.0%
	average delay (ms)	13	55	248
	max delay (ms)	66	193	363
	TCP thp. (Kbps)	161.38	17.15	4.33
BQ over 802.11e AC0	loss ratio	0.07%	1.9%	4.1%
	average delay (ms)	13	19	22
	max delay (ms)	58	70	72
	burst times	37	664	2830
	TCP thp. (Kbps)	163.13	59.43	69.08

The results in Table II shows that 802.11e can support VoIP better than 802.11b. When $M = 4$, the loss ratio of legacy 802.11e is zero. For BQ, 37 burst times result in a little higher loss ratio (0.07%), while it improves the TCP throughput by 2 Kbps. As M increases, BQ always achieves a good delay, the TCP throughput is much better than that of legacy 802.11e. As shown in Table II, for $M = 5$ and 6, BQ reduces the average delay by 65% and 91.1% respectively, and improves the TCP throughput by 246% and 1494%. We can see that BQ sacrifices a little packet loss ratio to obtain a lower delay, and the TCP throughput of BQ is always higher than that of legacy 802.11e.

To further increase the reliability of the transmission, we also tried to introduce a mechanism to acknowledge the broadcast. But even this kind of protocol consumption is not acceptable since the performance declines a lot. Most of the simulations are run over 802.11b, we do not run them again over 802.11g. That is because 802.11g just increase the transmission rate, the problems occur over 802.11b will also occur over 802.11g. We also tried to open CTS/RTS handshake, but the effort proves to be in vain. So we can conclude that the multihop environment is too tough for voice transmission, if we still focus in the frame work of IEEE 802.11, the problem cannot be perfectly solved.

D. BQ in grid topology

We also evaluate BQ scheme in 7×7 grid topology with GPSR and AODV. The distance between each neighboring nodes on dash lines is 100 meters. We test 4 scenarios (A,B,C,D) which have different CBR flows in the network. A: [A0- > G6, G6- > A0, G0- > A6, A6- > G0] ([source- > destination]) B: [E0- > E6, D0- > D6, G2- > A2, G3- > A3] C:[A0- > G6, G0- > A6, D0- > D6, G3- > A3] D:[E0- > E6, E6- > E0, D0- > D6, D6- > D0].

Figures 5 to 6 show that BQ scheme can greatly improve the performance of network in GPSR routing protocol.

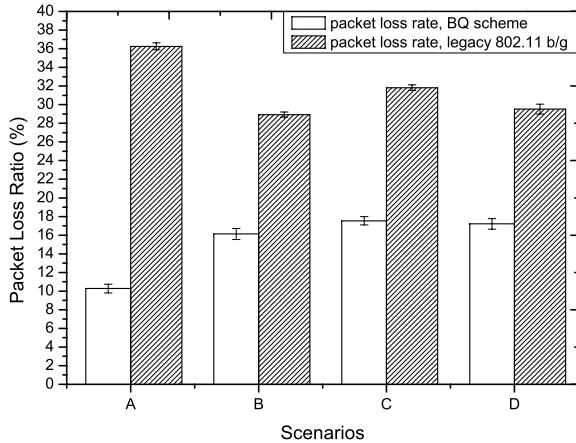


Fig. 5. Packet Loss Rate in Grid Topology for GPSR

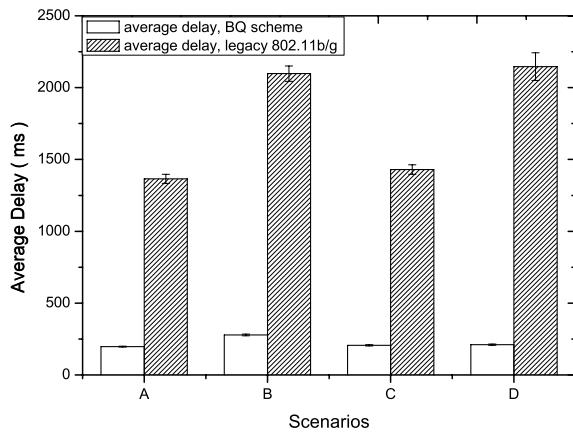


Fig. 6. Packet Delay in Grid Topology for GPSR

VI. CONCLUSION

In heavy load conditions, contention becomes the most serious problem of the performance. For a voice oriented network, it was filled with huge amounts of small packets which all contend for the shared network resources. To alleviate this situation and improve the network reuse, we first make the voice packet more efficient, say each packet gets higher information overhead ratio. Second is to reduce the protocol consumption include hand shaking and backoff time. We solve the first problem by packing several MAC packets together and just hold their MAC destination address. Thus the whole packet info-overhead ratio increased. To solve the second problem, we introduced burst broadcast. For small voice packets, 802.11 four way hand shake is too wasteful.

So broadcast becomes the most efficient way. Burst broadcast can also solve part of contention consumption that it reduces the total amount of DIFS, increase the efficiency of the channel and also prevents more contention. With out MAC layer ACK, Broadcast of course introduces more collision, but can reduce contention and delay, and improves background TCP throughput, thus it is a tradeoff between contention and collision, and can be used wisely.

VII. ACKNOWLEDGMENT

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