Multimedia Performance for IEEE.802.11 DCF RTS/CTS with Varying Traffic Conditions

R Razavi, M. Fleury, and S. Monaghan

Abstract— **Delay and jitter are important issues for real-time multimedia applications, particularly in conversational mode. Two causes of variable delay arise in IEEE 802.11's DCF RTS/CTS: packet arrival behavior at a transmitter's buffer; and subsequent access contention for the wireless channel. This paper provides simulated results for both delay and jitter behavior for varying traffic sources. Bursty and non-bursty traffic conditions are modeled.**

***Index Terms*—IEEE 802.11, bursty traffic. delay and jitter.**

I. INTRODUCTION

THE delay characteristics of the IEEE.802.11 [1] DCF in infrastructure mode have received considerable attention from the theorist, e.g. [2][3][4]. For example, the most recent of these contributions, [2], derives the access delay generating function and resulting distribution rather than the average delay, now taking into account dependencies between competing nodes. However, these analytical results [2][3][4] do not consider the combined delay that occurs as a result of queuing at a transmitter's buffer[1] *and* the delay from the time a packet reaches the head of the transmission queue until the instant when the packet is successfully received (or successfully acknowledged). If the packet arrival rate is not deterministic then the behavior of the overall delay distribution may change, as might its variance (jitter).

For real-time applications and real-time multimedia applications in particular the impact of delay and jitter is an important consideration. For video streaming, delay will affect the start up time, which ideally should be imperceptible. If mobile devices are involved then longer start-up delay requires larger play-out buffers, which in turn impinges on the memory power budget. For conversational services, the requirements for delay (and jitter) are more stringent, particularly for speech. For Voice-over-IP, work in [5] presented a set of user satisfaction levels depending on play-out delay, packet loss, and echo loudness. For example, with a moderate loudness rating of 55 dB, with less than 5% packet loss, a play-out delay of about 220 ms made most users satisfied and greater delays led to dissatisfaction. Again, [6] gives a table of tolerable levels of media synchronization levels is given. For example, the maximum tolerable skew for lip synchronization is about 80 ms.

However, our main concentration in this paper is on video. Observers note [7] that video streaming applications often do send a bursty stream, either for reasons of coding efficiency or when a non-real-time operating system falls behind its schedule and releases a packet burst. Bursty traffic requires a departure from Poisson statistics when modeling wireless networks under a video load [8]. However, no doubt for analytical convenience packet arrival rate may be modeled as a Poisson process [9]. In this paper, the Pareto probability density function (pdf) with varying shape factor is applied to packet inter-arrival times (PITs) in order to vary the input video traffic between classical Poisson and non-Poisson statistics. Traffic burstiness is also considered through a two-state Markovian model and its impact is contrasted with non-bursty traffic conditions.

A. IEEE 802.11 MAC

The IEEE.802.11 MAC Distributed Coordination Function (DCF) in which stations contend for access to the RF channel resolves conflicts by a Carrier Sense Multiple Access mechanism with Collision Avoidance (CSMA/CA). The RTS/CTS (Request To Send/Clear To Send) MAC extension to basic access seeks to minimize delay from collisions and out of range (hidden) stations [10][11]. If a station wants to transmit a data packet, it senses the medium. If the medium is free for a DCF Inter-Frame Spacing (DIFS) it begins transmitting an RTS frame. All other stations that hear this packet will set their Network Allocation Vector (NAV) for the duration of the transmission, including the time that it takes for the acknowledgement packet to be returned back to the sender. In fact, when the NAV is configured, a station knows that it should avoid transmission until the end of the NAV or a collision will occur on the shared RF channel.

When the base station receives the transmitted RTS frame, it replies with a CTS frame after a Short Inter-Frame Space (SIFS). Again the NAV will be configured by any station that hears the CTS packet but was unable to hear the RTS. If the CTS is not received, an RTS frame is retransmitted at some random time within a contention window. This eventuality arises because a vulnerable period[2] occurs between sending an RTS and successfully receiving a CTS frame during which

This work was supported by the EPSRC, UK under grant no. EP/C538692/1.The authors are with the University of Essex, Electronic Systems Engineering Department, Wivenhoe Park, Colchester CO 4 3SQ, Essex, U.K. (corresponding author M. Fleury at `phone: +44 (0)1206-872817; fax: +44 (0)1206-872900; e-mail: fleum@essex.ac.uk)

[1] Of course, there is a considerable queuing theory literature that separately considers queuing at the transmit buffer.

[2] One definition of the vulnerable period is (length of the RTS packet +length of the CTS packet) / channel rate + 2 × (propagation time to the base station + SIFS).

RTS/RTS and RTS/CTS collisions can occur. If a further collision occurs the contention window is doubled in size. The process continues until an RTS succeeds or a fixed number of binary exponential back-offs has occurred. Once RTS and CTS frames have been successfully exchanged, after an SIFS, a data packet is transmitted, with a data NAV being set by listening stations [11]. The transaction concludes, after an SIFS, with receipt of an acknowledgment frame (ACK) at the transmitter.

B. *Simulations*

As wireless systems are complex, it is not always possible [12] to capture all relevant features in an analysis. A simulator, which models the MAC layer, was designed specifically to extract delay values, resulting in an increase in simulation speed. Notice that the MAC implementation in the well-known ns-2 simulator is reported [2] to contain several bugs.

The default settings for the discrete event simulations are recorded in Table 1. The settings are for the 802.11b variant of the protocol using direct sequence spread spectrum (DSSS) modulation. In the simulations, not only the access delay but the queuing delay at the transmitter is accounted for. However, to isolate relevant results, the transmitter queue length was set to a size that avoided all packet loss. In practice buffer overflows may occur, which would decrease the delays reported herein but would also decrease the quality of service for a multimedia application. The simulations also include the (deterministic) service time to send a data packet, but do not include a further delay, as far as the transmitter is concerned, in receiving a final acknowledgment. This is because, for real-time services, once the data are received at the destination, processing can commence. The WLAN is assumed to be in infrastructure mode, with all transmissions sent to the base station. Scheduling time at the base station is neglected in the Section III's simulations.

To extend the reception range, reducing the problem of out of range stations [13], RTS and CTS packets along with headers are normally sent at the basic rate of 1 Mbps. Multiple stations contend for access to the base station, according to the RTS/CTS model set out in Section II.A. The RF channel was assumed by default to be error free, in order to isolate the impact of the MAC upon delay. Propagation delay was taken to be a constant value. The contention window size was set to the minimum allowed by the standard, while other settings were also according to the IEEE 802.11 standard.

C. *Non-bursty traffic model*

As a baseline for comparison, a deterministic packet arrival rate was simulated to determine the behavior of delay as various parameters were varied. The same traffic is output from all stations but binary exponential back-off removes any synchronization between arrivals, imposing its own distribution upon arrival rates onto the channel.

To judge the effect of varying the traffic conditions, a long-tailed Pareto distribution was imposed upon PITs at each station's transmitter buffer. To ensure that the sample distribution approaches the population distribution, the total number of packets simulated (Table 1) was 10×200000.

TABLE I DEFAULT PARAMETERS FOR THE SIMULATIONS

Number of stations	10
Data rate	5.5 Mb/s
No. of packets transmitted per run	200000
No. of runs per data point	10
Average distance to base station	100 m
Packet payload length	500 B
Queue discipline	FIFO
RTS length	160 bits
CTS length	112 bits
ACK	112 bits
Physical header	48 bits
MAC data header	224 bits
SIFS	10 µs
DIFS	50 µs

The mean value of a Pareto distribution is

$$\mu = \frac{a.x_M}{a-1}, a > 1 \tag{1}$$

when, a is the shape factor and x_M is the minimum possible value of x, the location. The shape factor, a, was also varied to change the variance, with larger values of a resulting in reduced variance according to (2):

$$\sigma^2 = \frac{x_M^2 a}{(a-1)^2 (a-2)}, a > 2 \cdot \tag{2}$$

D. *Bursty traffic model*

To model bursty packet arrivals at a transmitter station's buffer, a Markovian two-state model was employed. The mean arrival rate was kept constant over time to avoid changing the offered load, according to (3):

$$C = \frac{T_H.R_H + T_L.R_L}{T_H + T_L}, \tag{3}$$

where T_H is the time in the high state, R_H is the rate in the high state, T_L is the time in the low state, and R_L is the rate in the low state, while C is the desired constant packet arrival rate. Though the distribution of the PITs remained as Pareto, in order to simplify the interpretation of the results, the shape factor was set to 10, sharply reducing the variance, so that the distribution behaves almost 'as if' it were deterministic.

Now define

$$k = \frac{T_H}{T_L} \tag{4}$$

and

$$R = \frac{R_H}{R_L}. \tag{5}$$

From (3), this implies

$$C = \frac{R_L(R.k+1)}{k+1} \cdot \tag{6}$$

and

$$R_L = \frac{C.(k+1)}{R.k+1} \cdot \tag{7}$$

The other parameters are related by

$$R_H = R.R_L \quad (8)$$

$$T_H = kT_L \quad (9)$$

Moreover, the transition probability of being in a high state and remaining in it is defined as

$$P_{HH} = \frac{1}{1-T_H} \quad (10)$$

and similarly, the transition probability of being in a low state and remaining in it is defined as

$$P_{LL} = \frac{1}{1-T_L}. \quad (11)$$

As a result, the model is defined by three parameters: 1) R, which is a measure of the level of burstiness, 2) C which, as already mentioned, is the long term average of the input packet arrival rate, and 3) T_H , which is the mean time in the high state and can represent the scale of burstiness.

Unless otherwise stated, *C* is set at 90 packet/s per station, *k* = 0.2 by default, while *R* is variable.

II. RESULTS

A. Deterministic traffic

Delay measured is between a transmitter and a base station. Fig. 1 shows the overall delay versus the offered load (in packet/s per transmitter station). In this simulation, the default value of 90 packet/s was varied, while other settings remained the same. The arrival rate, λ, impacts on both the average number of packets in the buffer, *N*, and on channel load. From Little's theorem *N* = λ*W*, where *W* is the average per packet waiting time, implying in a completely general way that waiting time rises. The plot shows a rising delay but the mean delay remains acceptable for multimedia applications, even for higher arrival rates. The jitter is plotted in Fig. 2 and again values remain within acceptable bounds.

Packet length also affects delay. Fig. 3 shows a linear growth in delay and second order behavior for jitter. The behavior of delay with packet length is consistent with the analytical results in [14]. In our simulation, while packet length varied, the offered load was kept constant and moderate at 3.8 Mbps (including headers), which is the same as for a packet length of 500 B and an arrival rate of 90 packet/s.

B. Pareto traffic

If the configuration remains the same but the PIT distribution is switched to Pareto, then for lower packet arrival rates, the average delay and jitter remain within acceptable levels, Figs. 4 and 5. However, delay and jitter diverge for higher packet arrival rates (beyond about 100 packet/s per station). For high variance in PIT, *a* = 2.1, delay certainly becomes unacceptable, while jitter verges on the unacceptable.

C. Bursty traffic

In this Section, the two state Markovian model of burstiness from Section II.D is applied. Fig. 6 shows significant divergence in delay over a comparatively small range of packet arrival rates. At a value of *R* = 6, with the higher packet arrival rate being many times higher than the lower rate, with a mean arrival rate of 120 packet/s per station, delay is unacceptable and what is more continues to rise. Jitter is also unacceptable for the same settings.

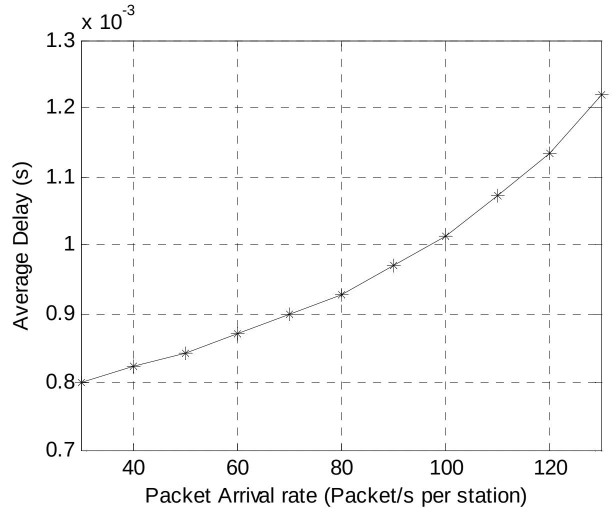

Fig. 1. Change in average delay as packet arrival rate is varied, PIT pdf deterministic, packet size = 500 B, transmission rate = 5.5 Mbps, no. of stations = 10.

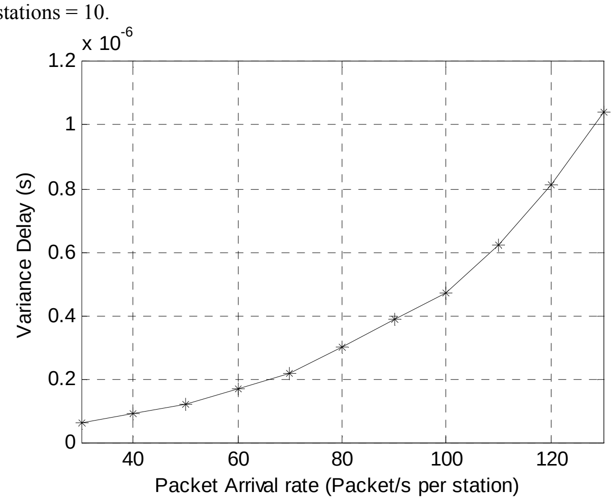

Fig. 2. Change in jitter as packet arrival rate is varied, PIT pdf deterministic, packet size = 500 B, transmission rate = 5.5 Mbps, no. of stations = 10.

Notice that a rate of 120 packet/s per station, with packet size 500 B, corresponds to a maximum of 5.1 Mbps with headers, whereas the transmission rate is 5.5 Mbps. Jitter climbs to high levels for moderate to high offered loads, Fig. 7, rising rapidly after a particular level of burstiness is reached. A plateau is reached, which certainly for a high offered load is undesirable.

Fig. 8 illustrates the changes in behavior as the burst scale varies, that is as time at the higher arrival rate increases. The level of burstiness remains constant throughout, with R = 5. The packet arrival rate is 90 packet/s per station, which is the same as for the deterministic and Pareto pdf simulations. Both delay and jitter rise almost linearly and verge on the unacceptable for a burst of 80 packets.

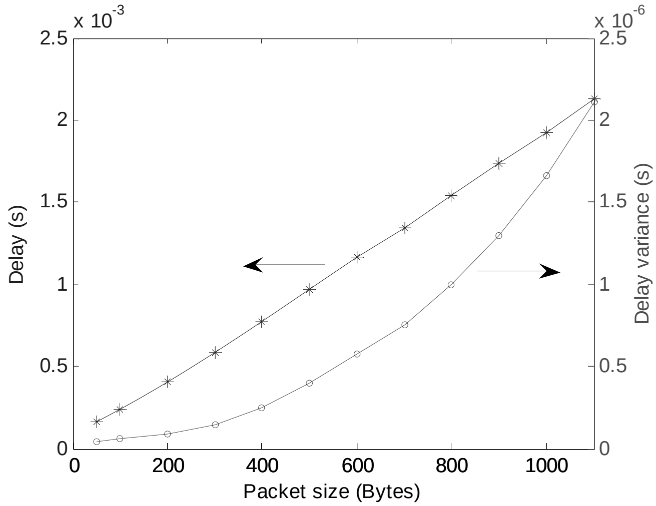

Fig. 3. Change in delay as packet length is varied, packet arrival rate 90 packet/s per station, PIT pdf deterministic, transmission rate = 5.5 Mbps, no. of stations = 10.

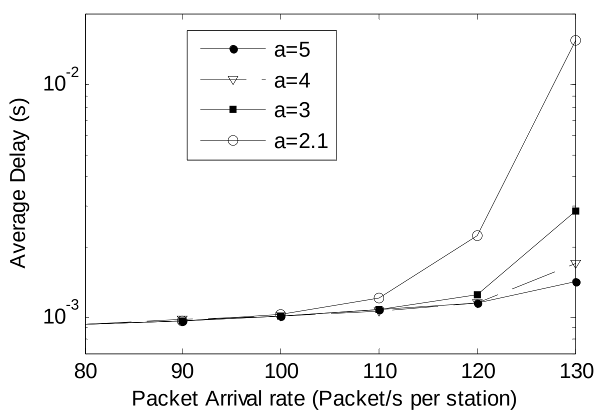

Fig. 4. Change in average delay as packet arrival rate is varied, PIT pdf Pareto, packet size = 500 B, transmission rate = 5.5 Mbps. (logarithmic vertical scale)

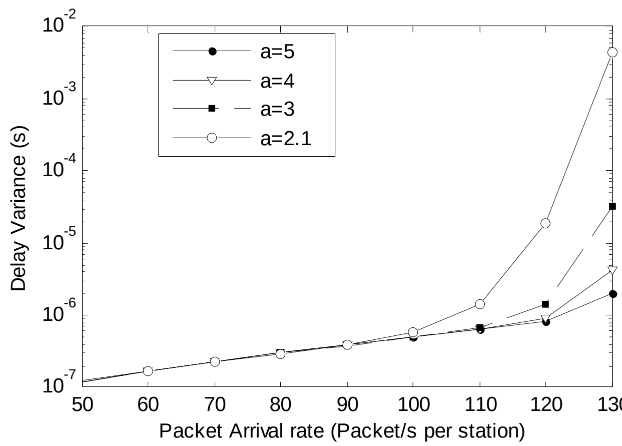

Fig. 5. Change in jitter as packet arrival rate is varied, PIT pdf Pareto, packet size = 500 B, transmission rate = 5.5 Mbps, no. of stations = 10 (logarithmic vertical scale).

III. CONCLUSION

For one-way real-time multimedia services potential delay determines the size of play-out and jitter buffers. For conversational services then these solutions are not necessarily available. This paper has found that certainly for heavily loaded IEEE 802.11b networks with DSSS, levels of delay and jitter may well become unacceptable, according to the scale (duration) of and level (rate of packet arrival) of packet bursts. The effect is compounded for larger packet sizes, which implies that video will be more affected than speech services. To reduce the burstiness of video traffic, smoothing [15] is possible and an appropriate scheme can be selected for this wireless LAN. Future work will also consider the effect of a noisy channel, as clearly this will also increase delay.

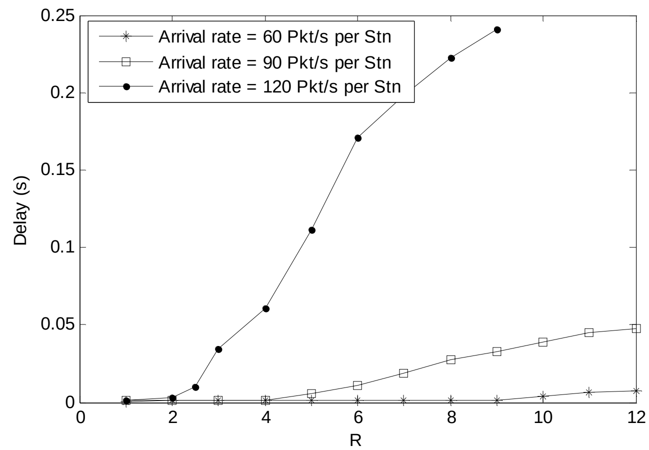

Fig. 6. Change in mean delay as the level of burstiness is varied, packet size = 500 B, transmission rate = 5.5 Mbps, no. of stations = 10, with burst scale *k* = 0.2.

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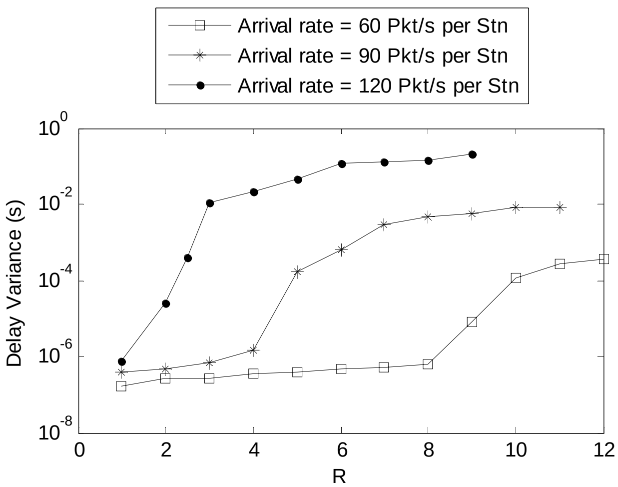

Fig. 7. Change in jitter as the level of burstiness is varied, packet size = 500 B, transmission rate = 5.5 Mbps, no. of stations = 10, with burstiness scale k = 0.2 (logarithmic vertical scale).

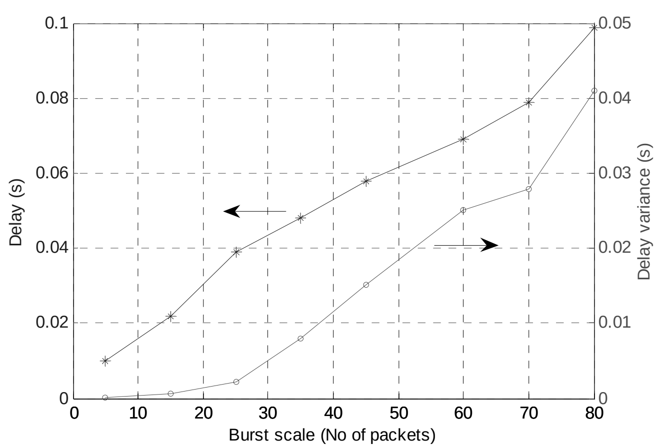

Fig. 8. Change in delay and jitter as the burst scale is varied, packet size = 500 B, transmission rate = 5.5 Mbps, packet arrival rate = 90 packet/s, R = 5, no. of stations = 10.