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Abstract: In this paper we present wired routers, for whom the sizing of buffers is an active research topic. The classical rule of thumb for sizing wired buffers is to set buffer sizes to be the product of the bandwidth and the average delay of the flows utilizing this link, namely the Bandwidth-Delay Product (BDP) rule. Surprisingly, however the sizing of buffers in wireless networks (especially those based on 802.11/802.11e) appears to have received very little attention within the networking community. Exceptions include the recent work in relating to buffer sizing for voice traffic in 802.11e WLANs, work in which considers the impact of buffer sizing on TCP upload/download fairness, and work in which is related to 802.11e parameter settings.

***Keywords**: WLAN, TCP/IP, eBDP, MAC* **.**

1. INTRODUCTION

The distribution of packet service times is also strongly dependent on the WLAN offered load. This directly affects the burstiness of transmissions and so buffering requirements. IEEE 802.11b support up to 11 Mbps, sometimes this is not enough – far lower than 100 Mbps fast Ethernet

In co-existing environment, the probability of frequency collision for one 802.11 frame vary from 48%-62%. The following are Disadvantages of existing model:

- Unaware of interference from/to other networks
- Weak security policy
- Poor performance (coverage, throughput, capacity, security)
- Unstable service
- Customer dissatisfaction

The sizing of buffers in wireless networks (especially those based on 802.11/802.11e) appears to have received very little attention within the networking community.

Exceptions include the recent work relating to buffer sizing for voice traffic in 802.11e [2] WLANs, work which considers the impact of buffer sizing on TCP upload/download fairness, and work in which is related to 802.11e parameter settings. Buffers play a key role in 802.11/802.11e wireless networks. To illustrate this, we present measurements from the production WLAN of the Hamilton Institute, which show that the current state of the art which makes use of fixed size buffers, can easily lead to poor performance.

We recorded RTTs before and after one wireless station started to download a 37MByte file from a web-site. Before starting the download, we pinged the access point (AP) from a laptop 5 times, each time sending 100 ping packets.

The RTTs reported by the ping program was between 2.6-3.2 ms. However, after starting the download and allowing it to continue for a while, the RTTs to the AP hugely increased to 2900-3400 ms. This reduction in delay does not come at the cost of reduced throughput, i.e., the measured throughput with the A* algorithm and the default buffers is similar.

In this paper, we consider the sizing of buffers in 802.11 /802.11e based WLANs. We focus on single hop WLANs since these are rapidly becoming ubiquitous as the last hop on home and office networks as well as in so called "hot spots" in airports and hotels, but note that the proposed schemes can be easily applied in multi-hop wireless networks.

Our main focus in this paper is on TCP traffic since this continues to constitute the bulk of traffic in modern networks (80–90% of current Internet traffic and also of WLAN traffic), although we extend consideration to UDP traffic at various points.

We propose the following advantages in this model:

- The reduction in network delay not only benefits UDP traffic, but also short-lived TCP connections
- Comes from easy maintenance, cabling cost, working efficiency and accuracy
- Network can be established in a new location just by moving the PCs.

2. IEEE 802.11 MAC

Measured distribution of the MAC layer service time when there are 2 and 12 stations active. It can be seen that the mean service time changes by over an order of magnitude as the number of stations varies. Observe also from these measured distributions that there are significant fluctuations in the service time for a given fixed load. This is a direct consequence of the stochastic nature of the CSMA/CA contention mechanism used by the 802.11/802.11e MAC.

Even this property provides the flexibility for practical usage, the false alarm of the rightful ownership verification is essentially existed since the resolution requirement for the transformed coefficients can he less demanded than other applications like compression or encryption. Image watermarking is the process of inserting an image called watermark in another image called cover image.

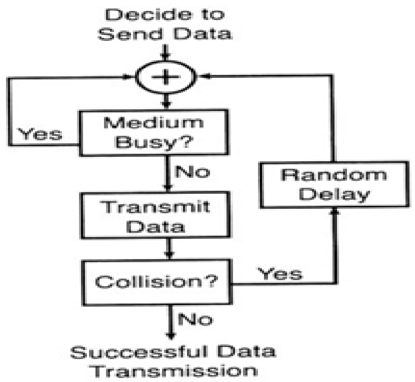

Figure 1 MAC functional flow

TCP/IP packet in 802.11

Consider a WLAN consisting of n client stations each carrying one TCP upload flow. The TCP ACKs are transmitted by the wireless AP. In this case TCP ACK packets can be easily queued/dropped due to the fact that the basic 802.11 DCF ensures that stations win a roughly equal number of transmission opportunities. Namely, while the data packets for the n flows have an aggregate n/(n + 1) share of the transmission opportunities the TCP ACKs for the n flows have only a 1/(n+1) share. Issues of this sort are known to lead to significant unfairness amongst TCP flows but can be readily resolved using 802.11e functionality by treating TCP ACKs as a separate traffic class which is assigned higher priority. With regard to throughput efficiency, the algorithms in this paper perform similarly when the DCF is used and when TCP ACKs are prioritized using the EDCA as in. Per flow behavior does, of course, differ due to the inherent unfairness in the DCF and we therefore mainly present results using the EDCA to avoid flow-level unfairness.

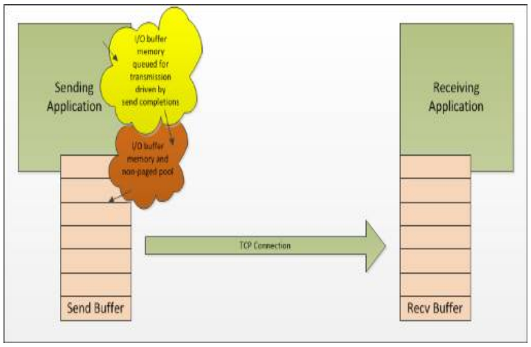

Figure 2 TCP/IP Packet Transfer

IEEE 802.11e Simulation

In this paper is on TCP traffic since this continues to constitute the bulk of traffic in modern networks (80–90% of current Internet traffic and also of WLAN traffic), although we extend consideration to UDP traffic at various points during the discussion and also during our experimental tests.

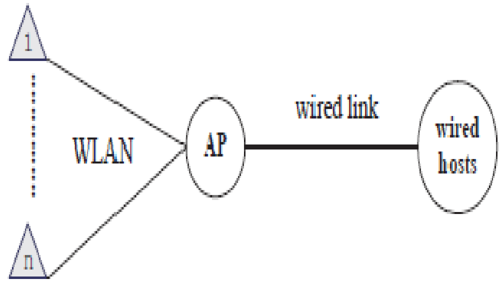

Figure 3 WLAN topology used in simulations Wired link bandwidth 100Mbps.

Compared to sizing buffers in wired routers, a number of fundamental new issues arise when considering 802.11-based networks. Firstly, unlike wired networks, wireless transmissions are inherently broadcast in nature which leads to the packet service times at different stations in a WLAN being strongly coupled. For example, the basic 802.11 DCF ensures that the wireless stations in a WLAN win a roughly equal number of transmission opportunities, hence, the mean packet service time at a station is an order of magnitude longer when 10 other stations are active than when only a single station is active.

Consequently, the buffering requirements at each station would also differ, depending on the number of other active stations in the WLAN

In this paper, in addition to extensive simulation results we also present experimental measurements demonstrating the utility of the proposed algorithms in a test bed located in office environment and with realistic traffic. This latter includes a mix of TCP and UDP traffic.

Traffic Mix, Adaptive Limit Tuning (ALT)

We configure the traffic mix on the network to capture the complexity of real networks in order to help gain greater confidence in the practical utility of the proposed buffer sizing approach.

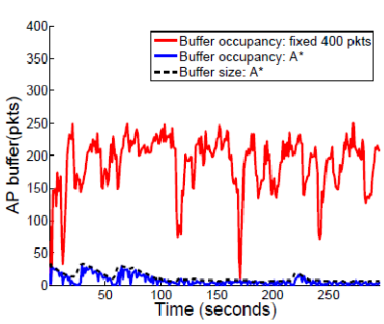

Figure 4 Time Histories of the Buffer Size of proposed model

Fig.4 shows example time histories of the buffer size and occupancy at the AP with a fixed buffer size of 400 packets and when the A* algorithm is used for dynamic buffer sizing. Note that in this example the 400 packet buffer never completely fills. Instead the buffer occupancy has a peak value of around 250 packets. This is due to non-congestive packet losses caused by channel noise which prevent the TCP congestion window from growing to completely fill the buffer. Nevertheless, it can be seen that the buffer rarely empties and thus it is sufficient to provide an indication of the throughput when the wireless link is fully utilized.

3. PROPOSED ALGORITHM

We begin by considering a simple adaptive algorithm based on the classical BDP rule. Although this algorithm cannot take advantage of statistical multiplexing opportunities, it is of interest both for its simplicity and because it will play a role in the more sophisticated A* algorithm developed in the next section.

As noted previously, and in contrast to wired networks, in 802.11 WLANs the mean service time is generally time varying (dependent on WLAN load and the physical transmit rate selected by a station). Consequently, there does not exist a fixed BDP value.

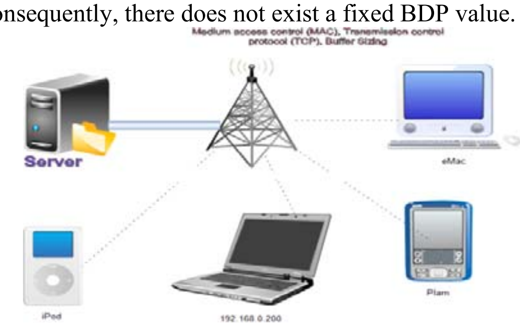

Figure 5 Proposed Architecture

However, we note that a wireless station can measure its own packet service times by direct observation, i.e., by recording the time between a packet arriving at the head of the network interface queue ts and being successfully transmitted te (which is indicated by receiving correctly the corresponding MAC ACK). Note that this measurement can be readily implemented in real devices, e.g. by asking the hardware to raise an interrupt on receipt of a MAC ACK, and incurs only a minor computational burden. Averaging these per packet service times yields the mean service time Tserv. To accommodate the time-varying nature of the mean service time, this average can be taken over a sliding window. In this paper, we consider the use of exponential smoothing

So, to isolate signal discontinuities one would like to have some short basis functions. In order to obtain detailed frequency analysis, one would like to have some very long basis functions. i.e. wavelet analysis provides immediate access to information that can be obscured by other time frequency methods such as Fourier analysis. Wavelet analysis has begun to play a serious role in a broad range of applications including signal processing, data and image compression, solution for partial differential equations, modeling multiscale phenomena and statistics etc. It provides a systematic way to represent and analyze multiscale structure.

It also provides a systematic and universal representation for wide classes of functions. Multiscale representation is a representation selected for the convenience of the analyst, because it often provides an efficient representation of information for storage, calculation or communication of information.

Tserv(k + 1) = (1 − W)Tserv(k) + W(te − ts) to calculate a running average since this has the merit of simplicity and statistical robustness (by central limit arguments). The choice of smoothing parameter involves a trade-off between accommodating time variations and ensuring the accuracy of the estimate – this choice is considered in detail later.

Given an online measurement of the mean service time Tserv, the classical BDP rule yields the following eBDP buffer sizing strategy. Let Tmax be the target maximum queuing delay.

Noting that 1/Tserv is the mean service rate, we select buffer size QeBDP according to QeBDP =min(Tmax/Tserv,QeBDP max) where QeBDP max is the upper limit on buffer size. This effectively regulates the buffer size to equal the current mean BDP. The buffer size decreases when the service rate falls and increases when the service rate rises, so as to maintain an approximately constant queuing delay of Tmax seconds. We may measure the flows' RTTs to derive the value for Tmax in a similar way to measuring the mean service rate, but in the examples presented here we simply use a fixed value of 200ms since this is an approximate upper bound on the RTT of the majority of the current Internet flows.

Algorithm Drop tail operation of the eBDP algorithm.

Set the target queuing delay T_{max}.
Set the over-provision parameter c.
for each incoming packet p **do**
  Calculate $Q_{eBDP} = min(T_{max}/T_{serv} + c, Q_{max}^{eBDP})$ where T_{serv} is from MAC Algorithm 2.
  if current queue occupancy $< Q_{eBDP}$ **then**
    Put p into queue
  else
    Drop p.
  end if
end for

Algorithm MAC operation of the eBDP algorithm.

Set the averaging parameter W.
for each outgoing packet p **do**
  Record service start time t_s for p.
  Wait until receive MAC ACK for p, record service end time t_e.
  Calculate service time of p: $T_{serv} = (1 - W)T_{serv} + W(t_e - t_s)$.
end for

Figure 6 Algorithm for proposed model

We therefore modify the eBDP update rule to QeBDP = min(Tmax/Tserv +c,QeBDP max) where c is an over provisioning amount to accommodate short-term fluctuations in service rate. Due to the complex nature of the service time process at a wireless station (which is coupled to the traffic arrivals etc at other stations in the WLAN) and of the TCP traffic arrival process (where feedback creates coupling to the service time process), obtaining an analytic value for c is intractable. Instead, based on the measurements in Fig. 3 and others, we have found empirically that a value of c = 5 packets works well across a wide range of network conditions. Pseudo-code for eBDP is shown in Algorithms 1 and 2.

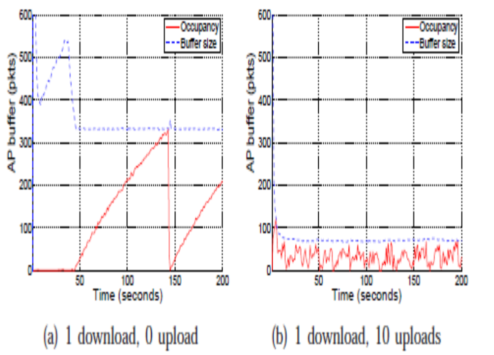

Figure 7 Histories of buffer size and buffer occupancy with the eBDP

The effectiveness of this simple adaptive algorithm is illustrated in Fig. 7. Fig. 7(a) shows the buffer size and queue occupancy time histories when only a single station is active in a WLAN while Fig. 7(b) shows the corresponding results when ten additional stations now also contend for channel access. Buffer sizes of 330 packets and 70 packets, respectively, are needed to yield 100% throughput efficiency and eBDP selects buffer sizes which are in good agreement with these thresholds.

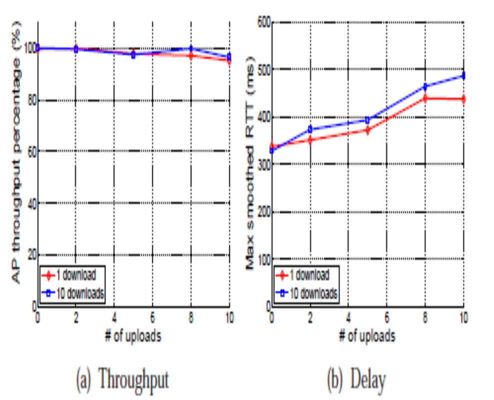

Figure 8 Performance of the eBDP algorithm as the number of upload flows

In Fig. 8 we plot the throughput efficiency (measured as the ratio of the achieved throughput to that with a fixed 400-packet buffer) and max smoothed RTT over a range of network conditions obtained using the eBDP algorithm.

It can be seen that the adaptive algorithm maintains high throughput efficiency across the entire range of operating conditions.

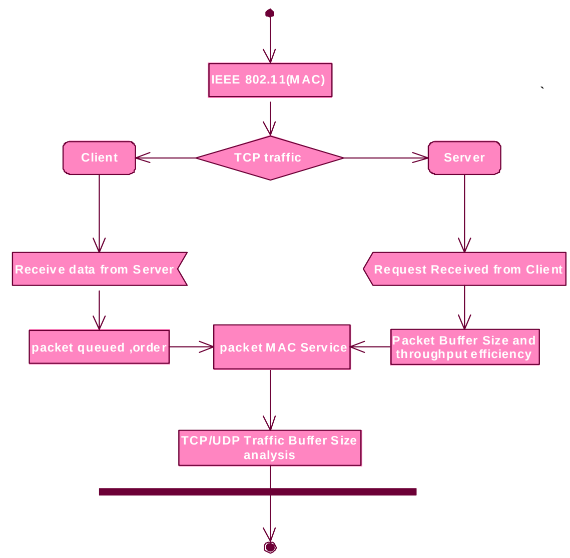

Figure 9 UML Activity Diagram

This is achieved while maintaining the latency approximately constant at around 400ms (200ms propagation delay plus Tmax = 200ms queuing delay) – the latency rises slightly with the number of uploads due to the over-provisioning parameter c used to accommodate stochastic fluctuations in service rate.

4. RESULTS & CONCLUSIONS

Fig. 4 shows example time histories of the buffer size and occupancy at the AP with a fixed buffer size of 400 packets and when the A* algorithm is used for dynamic buffer sizing. Note that in this example the 400 packet buffer never completely fills. Instead the buffer occupancy has a peak value of around 250 packets. This is due to non-congestive packet losses caused by channel noise which prevent the TCP congestion window from growing to completely fill the buffer. Nevertheless, it can be seen that the buffer rarely empties and thus it is sufficient to provide an indication of the throughput when the wireless link is fully utilized. We observe that while buffer histories are very different with a fixed size buffer and the A* algorithm, the throughput is very similar in these two cases. One immediate benefit of using smaller buffers is thus a reduction in network delays. The measured delays experienced by the UDP flows sharing the WLAN with the TCP traffic can be calculated.

It can be seen that for STA 8 both the mean and the maximum delays are significantly reduced when the A* algorithm is used.

This potentially has major implications for time sensitive traffic when sharing a wireless link with data traffic. Note that the queuing delays from STA 7 are for traffic passing through the high-priority traffic class used for TCP ACKs, while the measurements from STA 8 are for traffic in the same class as TCP data packets. For the offered loads used, the service rate of the high-priority class is sufficient to avoid queue buildup and this is reflected in the measurements.

We consider the sizing of network buffers in 802.11 based wireless networks. Wireless networks face a number of fundamental issues that do not arise in wired networks. We demonstrate that the use of fixed size buffers in 802.11 networks inevitably leads to either undesirable channel underutilization or unnecessary high delays.

We present two novel buffer sizing algorithms that achieve high throughput while maintaining low delay across a wide range of network conditions. Experimental measureme -nts demonstrate the utility of the proposed algorithms in a real environment with real traffic.

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