Fairness of IEEE 802.11 Distributed Coordination Function for Multimedia Applications

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Abstract - The Distributed Coordination Function (DCF) is the mandatory, and most widely used, access scheme in the IEEE 802.11 standard. Efficiency in utilizing wireless medium and fairness in respect to wireless stations and the traffic generated by them, have always been considered two of DCF's main features. In this paper, we examine the fairness of DCF in allocation of bandwidth to heterogeneous multimedia traffic, especially for overloaded systems. Results show that applications with certain traffic patterns, and queuing effects on heavily loaded stations, could disrupt fair allocation of bandwidth among different multimedia streams sharing the same station, as well as among stations. Among many others, one serious consequence of this can be significant asymmetry in quality of service between downlink and uplink directions in bi-directional multimedia sessions.

***Keywords* – DCF, Fairness, IEEE 802.11 Wireless LAN, Multimedia Applications.**

I. INTRODUCTION

WIRLESS computing, facilitated by deployment of wireless LANs, has gained widespread acceptance in recent years. It provides users with relatively high bandwidth wireless data connectivity. It also benefits from operating in the unregulated ISM band, which enables low cost wireless connectivity.

With the advent of the IEEE 802.11 standard for Wireless LANs, the IEEE 802.11 Medium Access Control (MAC) protocol [1] has been defined. Two coordination functions are specified in 802.11: the Distributed Coordination Function (DCF) is a basic multiple access technique utilizing Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA); while the Point Coordination Function (PCF) is a polling scheme, providing more explicit control over access of wireless stations to transmission medium.

It is a popular belief that DCF provides fair treatment to all users. This is often understood as equal allocation of bandwidth to all stations or, in some other cases, allocation of bandwidth to stations in proportion to the traffic they generate under lightly loaded conditions, and fair (equal) degradation of bandwidth available to each station under overload conditions. Simple analysis of the principles on which the 802.11 DCF is based reveals that neither of these popular beliefs is correct, and that the fairness of DCF should be understood as statistically equal opportunity to commence transmission of a frame among all stations that have a frame ready to transmit when the transmission medium has become available. Because of that, the relative traffic patterns of different traffic streams have a profound impact on the relative allocation of bandwidth to stations and traffic streams. In addition, under overload conditions, transmit buffers of some stations may become full, further distorting the allocation of transmission opportunities (and bandwidth) to different traffic streams.

Voice has traditionally been the primary driver of the growth in cellular wireless networks. However, voice and other multimedia applications are also becoming the dominating type of traffic in wireless LANs. Unlike voice-only systems (e.g. 2nd generation cellular networks), wireless LANs must be able to handle heterogeneous traffic with varying characteristics and quality of service (QoS) requirements. Some of the traffic types may be delay-sensitive (e.g. voice, video conferencing) while others may be less sensitive to delay but more sensitive to errors (e.g. email, file transfers). With these requirements in mind, and the current understanding of allocating transmission opportunities in DCF, it is necessary to understand how DCF will behave in respect to multimedia traffic streams. To the best of our knowledge, such observations have not been reported elsewhere.

A. Related Work

Many papers have studied the performance of DCF (e.g. [6][10][13]). Issues of fairness have been investigated for bi-directional applications among different wireless stations. Among issues of interest in these studies are location-dependent contention, trade-off between optimising channel utilization and achieving fairness, decentralized control and rate compatibility (also called Link Adaptation). Fairness has also been mentioned in reports on performance evaluation of IEEE 802.11 WLANs [2][5][6]. In [7], an attempt is made to achieve fairness by deriving an appropriate contention resolution algorithm. In [4], authors concentrate on experimental evaluation of 802.11b, and discuss fairness issues with regards to rate compatibility, namely the effect of the discrete set of supported signalling rates in 802.11b on the fairness in bandwidth allocation. Work reported in [3] proposes a priority-based MAC scheme (modified DCF) to achieve weighted fairness among multiple traffic flows, while maximising the wireless channel utilisation. In [2], the authors study throughput and fairness properties of the IEEE802.11 DCF, specifically the impact of hidden terminals and the capture effect. In [12], issues of fairness are also addressed for the scenario of overlapping Basic Service Sets (BSSs), where geographically co-located Wireless LANs share the radio channel. However, none of the papers mentioned above have analysed fairness with respect to multimedia streams, especially in a highly loaded and/or overloaded system.

B. Our Observations

Fairness is normally tested among two or more stations in a BSS, with identical offered load, and within the capacity boundaries of the MAC protocol. Under such conditions, fair (proportional) bandwidth allocation is expected. However, if stations transmit traffic streams generated by different applications, with more than one application serviced by a given station at a time, the packet generation rates and packet sizes differ from station to station and from one traffic stream to another. Under such conditions, the popular expectation of fair allocation of bandwidth to different traffic streams needs to be modified, especially if it is possible for the entire system or selected stations to become overloaded. In wireless systems featuring QoS control, admission control would normally be implemented for multimedia (e.g. voice and video) sessions [8], thus preventing overload. However, the best effort traffic is normally not fully controlled, in order to achieve good utilization of available bandwidth, and therefore it will be possible for the network to experience high load conditions. In this paper, we present some observations related to the issues mentioned here.

When the system using DCF is underloaded, stations and applications can always get the resources they require. However, when the system is overloaded, the key question is whether all stations or applications would suffer fairly the degradation in resources available. Observations resulting from simple experiments lead to a conclusion that under overload different traffic streams may suffer widely different levels of degradation in allocated resources. As an example, we can consider a station transmitting multiple streams of traffic. Let’s assume that one of these is a stream of frequent but small frames (e.g. as in the case of Voice-over-IP traffic). We observe that such traffic streams, characterised by frequent requests for access to the transmission medium, tend to dominate the use of available radio resources. This may turn into nearly exclusive occupation of radio resources under severe overload, if such streams compete for available buffer space in overloaded transmit buffers against streams producing infrequent but large frames of data.

Effects such as the domination of some streams over others described above, can also severely impact on the symmetry of resources allocated to bi-directional streams. Also, in a typical infrastructure network, the Access Point is expected to transmit relatively high volume of downlink traffic (and many separate streams) to other wireless stations. Some of these streams may be severely disadvantaged due to the effects described above.

C. *Outline of the Paper*

The short study described in this paper concentrates on the fairness of DCF in scenarios featuring multiple streams of multimedia traffic, via analysis of simulation results. The paper is organized as follows: Section II gives a brief overview of IEEE 802.11 Wireless LAN DCF. Section III gives information on the simulated environment and simulation setup. Section IV presents observations drawn from the results of simulations. In Section V, further discussion on some aspects of fairness is given. Conclusions, as well as comments on future work, are given in Section V.

II. IEEE 802.11 WLAN DCF AND FAIRNESS

The IEEE 802.11 DCF is based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). Before a station transmits, it senses the wireless medium to determine if it is idle. The transmission only proceeds if the medium has been sensed idle for a specified interval. Otherwise, the station waits until the end of the transmission in progress to seek the transmission opportunity again. When the medium becomes idle, a station has to perform a random back-off procedure (i.e. wait for a random time interval, to minimise the probability of collision with another station that has sensed the medium free).

The DCF adopts a slotted binary exponential backoff mechanism to select the random backoff interval. The random number is drawn from a uniform distribution over interval [0, CW-1], where CW is the Contention Window size with initial value of CW_*min*. After a successful transmission, the value of CW is reset to CW_*min*. Every station involved in the back-off procedure decrements its back-off counter as long as the medium is sensed idle. If the counter has not reached zero and the medium becomes busy again, the station freezes its counter until the medium becomes free again. Once the counter reaches zero, the station starts its transmission. Therefore, if the network is busy and there are several stations with a frame ready to transmit, the time each of the stations contending for access waits until transmission of a frame is, on average (over a number of transmissions), equal. This is where the notion of fairness has been derived from. In other words, when stations have frames in transmit buffers ready for transmission, each of these stations is given, on average, the same level of transmission opportunity. The numbers of frames sent by these stations during a certain period of time are (statistically) equal, only if all involved stations have packets in transmit queues during the period of time in question.

If all packets sent as described above are of the same size and have statistically comparable inter-arrival patterns, all stations can be expected to enjoy an equal allocation of bandwidth. However, packets generated by multimedia applications may have different arrival patterns and sizes. Under such conditions, it is important to know if proportional fairness in bandwidth allocation could still be achieved among different multimedia streams sharing the same station, and among stations with different traffic loads. Furthermore, for an overloaded system, the key question is whether all stations and/or application streams would suffer fairly the degradation of allocated resources.

III. THE SIMULATION ENVIRONMENT

In order to analyse the fairness of DCF in respect to multimedia streams, we have developed a simple simulation model of an IEEE 802.11b network, using the OPNET discrete event simulation package. The OPNET model includes traffic sources, the 802.11 MAC protocol, and characteristics of the 802.11 physical layer and radio medium (e.g. BPSK bit error rate model, free space path loss model). Several assumptions have been made for simulation experiments.

- The effect of 'hidden terminals' is not accounted for in the simulations, i.e. all nodes are located within hearing distance of every other node.
- The RTS/CTS scheme is not used in our simulations. It is not necessary for experiments aiming at observing fairness of DCF, in an environment free of hidden terminals.
- Only one BSS is simulated, therefore no interference from other BSSs is considered.
- The transmit buffer available in each station is finite; when the transmit buffer fills, all newly generated packets are dropped. Each station maintains only one buffer, common to all application streams serviced by the station.

The values of simulation attributes used in our experiments are provided in Table 1. In the table, the value of Physical Characteristics (DSSS Long) stands for Direct Sequence Spread Spectrum (DSSS) with long preamble; this scheme adopts 144 bits for the physical layer preamble and 48 bits for the physical layer header. The buffer size specifies the maximum length of the higher layer data arrival buffer. Each station maintains one buffer; once the buffer limit is reached, data arriving from the higher layer will be discarded until some packets are removed from the buffer.

Attributes	Value
Data Rate (Mbps)	11
Fragmentation Threshold (bytes)	2304
Physical Characteristics	DSSS Long
Buffer Size (bits)	2048000
DSSS preamble (bits)	144
DSSS header (bits)	48
slot_time (us)	20
SIFS_time (us)	10
Cw_min	31
Cw_max	1023

Table 1 Simulation attributes used in experiments

The performance measurements of interest in our analysis include:

- Throughput: in our simulations, it represents the total number of bits (in bits/sec) handed up from 802.11 MAC layer to higher layers;
- Load: it represents the total number of bits (in bits/sec) submitted to 802.11 MAC layer by higher layers.
- Data dropped: it is the total size of higher layer data packets (in bits/sec) dropped by the 802.11 MAC; it is due either to overflow of MAC layer buffer, or failure of transmission repeated 7 times.

A. *Multimedia Traffic Model*

In reality, Internet traffic is generated by many kinds of traffic sources such as interactive voice and video, audio and video download, and interactive or bulk data. In our simulation, four different traffic sources are modelled: interactive voice, audio download, video-conferencing and bulk data transfer. They are assumed here to represent a range of multimedia applications with different inter-arrival rates and packet sizes. Traffic generated by each user is simulated by independent ON/OFF sources, as in [9][13], with parameters listed in Table 2.

As shown in Table 2, voice packets have fixed size of 92 bytes and arrive every 20ms (in each stream). Audio frames are much larger and arrive less frequently, with a uniform distribution. The packet arrival process for a video-conference source consists of an ON-state and a silent OFF-state. In the ON-state, packets have constant interarrival interval and are large in size. The bulk data source is the most aggressive; it has an exponential distribution of interarrival times with high average frequency of arrivals, and very large packet size. Therefore it may consume large proportion of bandwidth. Another stream is also simulated, acknowledgements to the bulk data packets (Data Ack); it follows the same exponential distribution of interarrival times with the mean of 5ms, and features very small frames of 64 bytes.

Application	Rate (Kbps)	On/Off Time On(ms)	Off(ms)	Inter-arrival time (ms)	Pkt Size (bytes)
Voice	36.8	always on		20	92
Audio	52.0	always on		Uni(8.3,250)	815
Audio Ack	2.5	always on		Uni(8.3,250)	40
Video	1410.0	12	88	1	1464
Data	3680.0	always on		Exp (5)	2300
Data Ack	102.0	always on		Exp (5)	64

Table 2 Parameters of Multimedia Sources

B. *Network Topology*

Simulations cover the time span of 30 second, including period I (the first 10 seconds) and period II (the remaining 20 seconds). Data Station (sending the bulk data) becomes active only from the start of period II. This has been designed to allow observation of the effect of activation of high volume data stream on other streams. As mentioned earlier, we simulate only a single BSS with one Access Point (AP). As shown in Figure 1, the BSS includes the following stations:

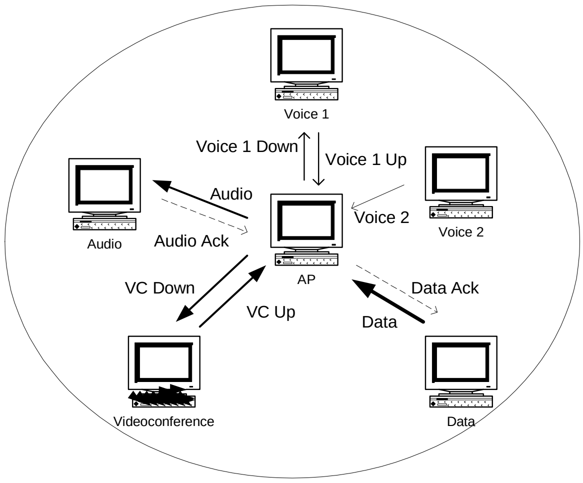

Figure 1 Simulated network topology

- Voice 1 Station: two-way interactive voice, active from the start of simulation period I.
- Voice 2 Station: one-way voice (to Access Point), active from the start of simulation period I.
- Audio Station: audio download from Access Point, with Ack feedback returned up-link to the Access

Point, active from the start of simulation period I

- Videoconference (VC) Station: two-way videoconference application, active from the start of simulation period I.
- Data Station: Bulk data transfer up-link to Access Point, with Ack returned down-link; active from the 10th second of simulation (i.e. the start of simulation period II)
- Access Point: In the simulation period I, it sends voice, audio and VC while also receiving the VC, voice and audio acknowledgments. In the simulation period II, it also receives the bulk data and sends bulk data acknowledgments.

IV. SIMULATION RESULTS

A. *Peak Throughput with Variable Packet Sizes*

First, we look at the peak throughput of the 802.11b wireless LAN, in order to gain a clear understanding of the impact of system overload. As shown in Figure 2, during period I (the first 10 seconds) all stations and applications receive their required bandwidth (total of 2.982Mpbs). No data is dropped during this period, and we consider the network lightly loaded. During period II, the network becomes overloaded with the bulk data. The total bandwidth requirement reaches 6.764Mpbs while the system can only offer a maximum of approximately 5.6Mpbs.

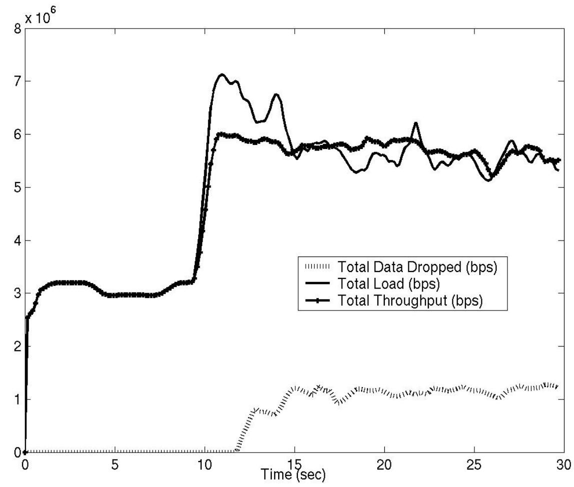

Fig. 2 Total Throughput, Load and Data Dropped

The 1.1Mpbs of data dropped is mainly due to the overflow of buffer(s) at one or more stations (note the delay in dropping as compared with the start of the bulk data transfer).

B. *Fairness in Respect to Different Stations*

In this section, we investigate the sharing of resources among different stations. As discussed in Section II, each station that has a frame to transmit will enjoy the same level of opportunity to transfer when the medium becomes free. In an overloaded system, there must be buffer overflow in some stations, even when the bandwidth requirements of other stations are satisfied. Such stations have, on average, more frames to send than the transmission opportunities they receive allow for. In other words, the average frame arrival rate in these stations is greater than the average frame transmission rate. Such overloaded stations not only fully consume their 'fair' share of transmission opportunities, but also 'excess' opportunities left over by other, less busy stations (i.e. stations that may, at times, have their transmit buffer empty due to less frequent arrivals of frames from traffic sources).

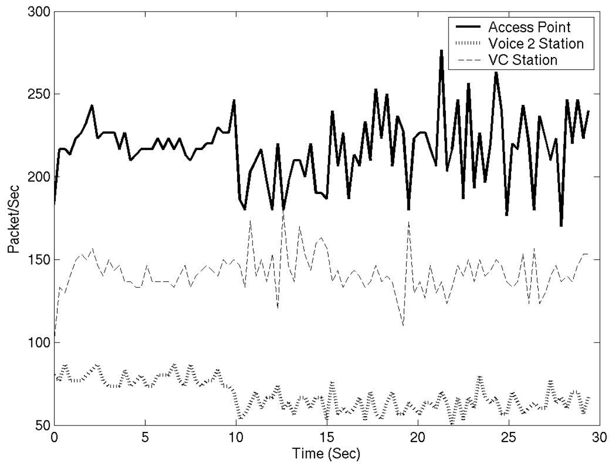

Fig. 3 Rate of Data Packets Sent by Stations

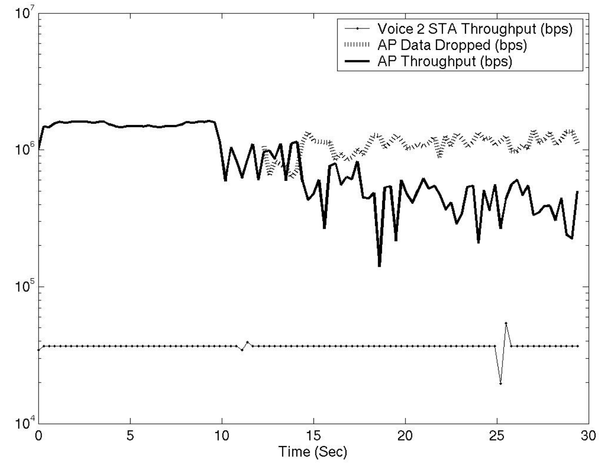

Fig. 4 Fairness between Voice 2 Station and AP

We first look at simulation results in terms of rates of data packets sent. This may be interpreted as the number of transmission opportunities taken (used). In Figure 3, we show appropriate results for AP, Voice 2 and VC stations (other results were left out for the sake of clarity/readability of the graph). AP sends traffic contributed by one videoconference, one audio, one data Ack (only Period II) and one voice stream, while Voice 2 station only needs to send one voice stream. According to the traffic patterns specified in Section III, frames of data Ack stream arrive with a much higher frequency than voice frames. At the same time, the frame sizes of audio and videoconference streams are much larger than the size of voice frames. Therefore, AP maintains a much larger buffer of frames ready to send than Voice 2 Station. Voice 2 Station uses its 'fair' share of transmission opportunities which are sufficient to achieve full required throughput with acceptable delay, while AP needs to consume more transmission opportunities to send its frames, thus using ‘excess’ opportunities left by other stations. In the simulation period II, when overload occurs, the AP cannot access transmission opportunities sufficient to satisfy its traffic, because the required ‘excess’ opportunities have to be shared with the bulk data station. As a result, AP experiences buffer overflow and starts dropping frames, while the Voice 2 Station can still have its requirement for transmission opportunities satisfied and thus maintain satisfactory throughput and delay, as shown in Figure 4. Results above (in Figure 3) illustrate that each station receives fair treatment in terms of transmission opportunities, even in an overloaded system, and that busy stations may also be able to share the ‘excess’ opportunities left by other stations. In the next sub-section, we further study the issues of fairness among streams serviced by the same busy station.

C. *Fairness among Application Streams in One Station*

When the system is lightly loaded, the sharing of resources among different application streams in one multi-stream station follows the same principles as those applicable to sharing of resources among different stations. However, in case of overload, the application streams in a multi-stream station may interact with each other in a different manner, highly dependent on packet arrival behaviours of the streams involved and the respective packet sizes.

We will observe the interaction between streams in an overloaded station on the example of Access Point. For the purpose of our discussion, we further divide the simulation period II into period IIa and period IIb. Period IIa is defined as the period from the start of system overload to the time when the transmit buffer at AP fills up. Period IIb follows. Streams serviced by the AP behave differently during periods IIa and IIb. During period IIa, the proportions between numbers of frames generated by different streams are reflected in the relative proportions of frames stored in the buffer. Because of the first-in-first-out service discipline, the sequence of frames in the buffer (and frames transmitted) still reflects the sequence of frames generated, as in period I. Therefore each stream drops almost the same percentage of its packets (i.e. suffers the same degree of degradation in allocated bandwidth). In Figure 4, the period IIa can be identified as spanning between 10 and 14 seconds. The relative drop in throughput experienced by the VC and voice streams transmitted by the AP during the period IIa can be seen in the Figure 5. Both streams experience a drop in throughput by approximately 50% (please note the logarithmic scale used in the graph).

However, during period IIb, the relative proportions between the numbers of frames in the buffer change. Because the buffer is full (or nearly full), most of the time it does not have enough room for the newly generated VC (large) frames, while the small voice frames could still fit in the buffer space freed up by a transmission of a frame. As a result, small but frequently generated frames will dominate the buffer, causing relatively larger packet (throughput) loss for streams with less frequent but large frames. In the case shown in Figure 5, the frames from Data Ack stream (64 bytes) are much smaller than videoconference frames (1504 bytes) and audio frames (815 bytes). Accordingly, we observe in Figure 5 that during the simulation period IIb the throughput of the videoconference stream decreases, while throughput of Data Ack and Voice streams increases. Not surprisingly, as shown in Figure 5, in period IIb (from 15th second approximately), Data Ack has almost full required throughput while the videoconference stream can only maintain approximately 25% of the throughput it requires, as compared to 50% in period IIa.

In order to verify our understanding of the buffer effect, we simulated the same scenario, this time with a much larger buffer (12144000bits); the results show that the larger buffer can only extend the duration of period IIa. Once the buffer fills up (or nearly fills up), stations and streams experience the same problems as described before.

We conclude that under overload conditions, traffic streams with short but frequent frames tend to dominate over other traffic streams within a multi-stream station. Moreover, a highly loaded station (typically, the Access Point will have more traffic to transmit than other stations) will, under overload conditions, tend to suffer more throughput degradation than other, more lightly loaded stations. In the extreme, down-stream traffic transmitted by the AP may suffer unacceptably high degradation of throughput, while up-stream traffic may still enjoy the required level of throughput.

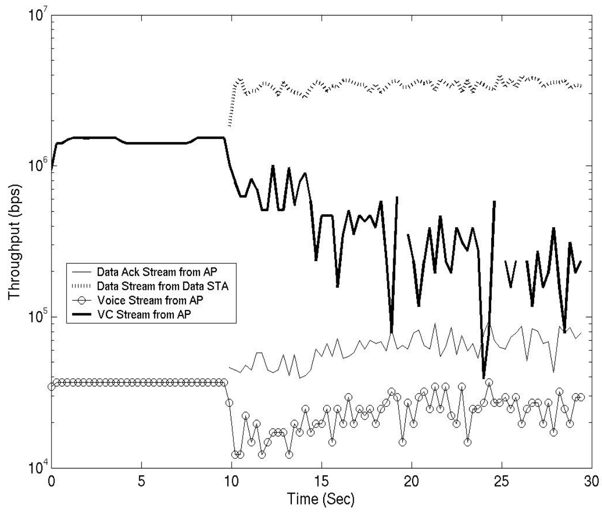

Figure 5 Multiple Streams Performance in AP

A good example of this unfair and asymmetric allocation of bandwidth, shown in Figure 5, is the throughput enjoyed by the data station, which requires 3.68Mpbs bandwidth for transmitting its bulk data stream. The station achieves nearly the required level of throughput, while the much lighter VC stream from the AP (overloaded with packets to transmit) suffers significant degradation of throughput.

V. DISCUSSION

The interaction of traffic streams described above has further effects on the traffic handling behaviour of the network.

A. Effect On Bi-directional Streaming

Bi-directional traffic streams (e.g. video or audio conference) often require symmetric throughput performance for both directions. In an "infrastructure" wireless LAN, these are the directions "to AP" and "from AP". The AP is almost always more loaded with frames to transmit than any other station in the network, since it transmits to all other stations. It is therefore almost inevitable that under heavy load (or overload), the downlink direction of communication will suffer more than the uplink one. This can be illustrated by the example of videoconference streams between the AP and the VC station in Figure 6. The VC, as has been described before, generates large and infrequent frames. Downlink VC traffic shares the resources available to AP with many other traffic streams (some of them with short and frequent frames) and therefore in the case shown in Figure 6 drops to approximately 20% of the required throughput when the network is overloaded. The uplink VC traffic, however, is the only traffic transmitted by the VC station, and because this station can capture sufficient share of total opportunities to transmit, the uplink direction of the VC enjoys sufficient level of throughput.

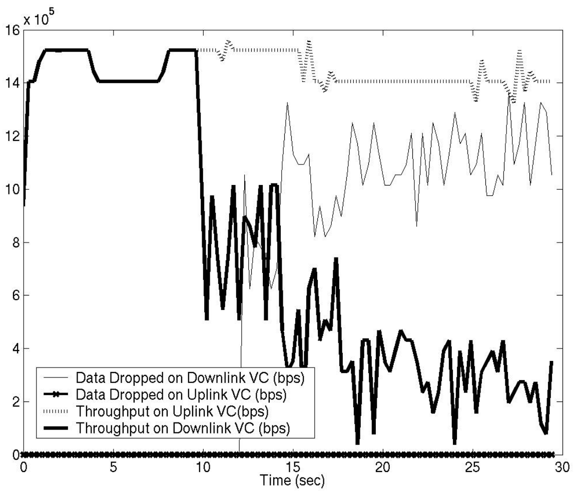

Fig. 6 Videoconference Performance

B. Effect On Uplink and Downlink

Internet traffic usually exhibits asymmetry between the uplink (lightly loaded) and downlink (highly loaded) directions. Since the AP has no more transmission opportunities than other stations (AP is only considered by DCF a "normal" station) this creates a significant imbalance between the transmission load on the AP and transmission load on other stations. This creates conditions under which the phenomena described before can cause significant degradation of quality of service for the downlink traffic (e.g. downloads of data in response to uplink requests), while the uplink traffic (e.g. requests) enjoys sufficient throughput.

VI. CONCLUSION AND FUTURE WORK

We have presented simulation results to illustrate the issues of fairness in respect to multimedia traffic over the IEEE802.11 DCF WLAN. In an underloaded system, the DCF ensures fair (i.e. proportional to the number of frames to send) allocation of resources to stations and traffic streams. However, when the system is overloaded, fair degradation of the level of service among stations and streams cannot be assumed, due to behaviour of limited-size buffers and individual traffic arrival patterns.

In overload conditions, traffic streams with light and frequent frames tend to dominate access to buffer space and radio resources available to a station, at the expense of traffic streams with larger and less frequent frames. Such problem is likely to affect performance of Access Points in an infrastructure network, as well as create asymmetry of throughput enjoyed by bi-directional applications.

These results show that DCF, although simple and "fair" in lightly loaded networks, fails to maintain its "fairness" under high load, especially when one station (usually the AP) has to transmit more than other stations. More sophisticated mechanisms are needed to control access of wireless stations to radio resources. Such mechanisms have to be capable of controlling admission of traffic streams to the resource-limited network, and to account for the greater transmission bandwidth requirements of some stations (e.g. APs) in the principles on which bandwidth allocation is based. In short, more explicit, sensitive to traffic characteristics, QoS control is needed in resource-limited wireless LANs.

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