EFFICIENT TRANSPORT OF IP FLOWS OVER IEEE802.11 WIRELESS MESH NETWORKS

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ABSTRACT

Wireless Mesh Networks (WMNs) based on the IEEE802.11 standards can be used as convenient replacements of wireline networks in the context of home networking to provide users with a variety of services inside the home context for their low cost and easiness of deployment. However, in order to effectively support multimedia as well as ordinary traffic through a WMN infrastructure, traffic control is generally needed as wireless communications, especially in mesh network configurations, can be affected by significant congestion and channel impairments problems. In this work we are focusing on performance assessment and enhancement in IEEE802.11 WMNs. Experimental results are presented using simulation models of IEEE802.11b technology obtained through the open source INET framework of OMNET++ simulator.

1. INTRODUCTION

IEEE 802.11 is a largely adopted technology for deployment of wireless local area networks (LANs) [1]. In this context, IEEE 802.11 is often configured to operate in the *infrastructure mode*, where a set of access points (APs) serve as communication hubs for mobile stations and provide entry points to the Internet, and the current role of IEEE 802.11 is limited to direct communications between mobile clients with a single AP.

In the context of home networking a more flexible network configuration using IEEE802.11 technology can include direct communication between some intrinsically coupled devices, such as a media server and a media renderer, in addition to multiple relay nodes for the whole wireless infrastructure in order to increase wireless coverage and reliability. In such context it is possible to use the IEEE 802.11 to form a *full wireless mesh network* (WMN) by means of two additional modes of operation. The *ad-hoc* mode can be used to establish a single-hop ad-hoc network where nodes communicate with each other directly without the use of APs. In addition, the wireless distribution system (WDS) mode allows forming *point-to-point AP relay links* where each AP acts also as a wireless relay node.

Such WMN, though enabling more flexible network configurations, does not generally possess satisfactory levels of QoS, for various reasons. The main are: *intrinsic unreliability of the wireless medium* along with the *potentially high number* of traversed hops that makes it difficult to provide bandwidth guarantees and *the contention based MAC scheme with IEEE802.11 WMNs*, i.e. CSMA/CA [2] operated with the DCF (Distributed Coordination Function), that poses serious challenges in the control of the end-to-end delay, as it does not assure time-bounded access [3, 4]. In order to support real-time multimedia communications in an IEEE802.11 WMN, one can compensate for the lack of effective traffic control strategy at the IP layer applying buffer management strategies. In addition, the alternative time-bounded PCF (Point Coordination Function) MAC scheme can be adopted along with the DCF based on CMSA/CA, to convey real-time traffic in particular.

In section II we introduce the theoretical models which we have used to study IEEE802.11 mesh networks. In section III we show experimental results obtained with the OMNET++ simulator.

2. TRAFFIC CONTROL AND CHANNEL ESTIMATION

In this work we present a control architecture based on Earliest Deadline First Packet Scheduler (EDFPS) [5] (depicted in Figure 2) applied to a WMN to support quality of service. The EDFPS consists of the following components:

- *A CLASSIFIER* separating IP packets into distinct flows;
- *FIFO queues* for storing packets of different service classes; queues dedicated to UDP traffic operate a delay jitter regulation by dropping packets reaching queuing delays beyond a certain threshold value, while, a RED/RIO[3] dropping strategy is set for queues storing TCP traffic.
- *LBs (Leaky Buckets)* used to limit bandwidth allocations to each service class;

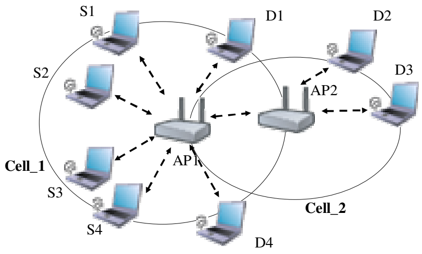

Figure 1 – WMN Scenario Example

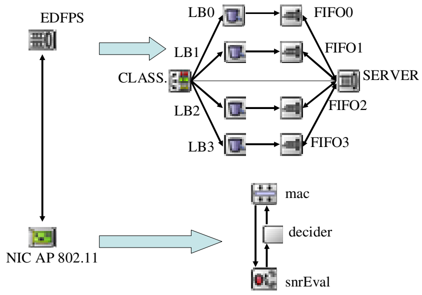

Figure 2 – traffic control architecture in an AP relay station

- *A SERVER PROCESS* implementing the EDF scheduling and PQ (Priority Queuing) scheduling.

We adopt an innovative closed loop algorithm based on channel bandwidth estimation in order:

- to render the scheduling process work-conservative and optimize use of the available bandwidth
- provide time bounded access for most critical applications conveyed on UDP flows.

Figure 1 shows a possible network configuration for a WMN, with two interconnected AP nodes heading two wireless cells, communicating with the WDS mode where the proposed architecture can be applied.

Overall capacity experienced in an IEEE 802.11b link [6] decreases from its nominal value on account of various factors, including: *channel impairments*, *overhead* and *link contention*. The considered network scenario consists of N nodes contending for resources: the contention adds further delay for medium contention and the layer overhead (MAC, IP and UDP/TCP headers) is also considered. The channel capacity of a IEEE802.11b link is bounded by

$$C(N) = C_{\max} \cdot \frac{t_{tr}}{T(N)}$$

where C_{max} is the nominal capacity of the link IEEE802.11 (e.g. equal to 11Mbps for IEEE802.11b), t_{tr} is a MAC PDU (protocol data unit) transmission and $T(N)$ is given by the following equation

$$T(N) = t_{tr} + t_{ov} + t_{cont}(N)$$

where t_{ov} is a fix overhead introduced by the CSMA/CA protocol and $t_{cont}(N)$ is an additional delay (an analytical formula can be found in [7, 8]) that has to be considered, corresponding to the time spent during contention among the N nodes. It can be shown that the maximum throughput for a multi-hop link (including headers overhead) which can be observed at application layer, i.e. B_{eff}, is given by

$$B_{eff}(L, BER, N_{\max}) = \frac{C(N_{\max}) \cdot L \cdot (1 - BER)}{2 \cdot (L + OV)}$$

where L is the length in bytes of the transmitted application PDU, BER the average bit error rate experienced along the multi-hop path and N_{max} the biggest number of contending nodes within each hop.

3. EXPERIMENTAL RESULTS

The simulation tool used to validate our solution is OMNET++ for its support of wireless channel and queues management. As far as the wireless link is concerned, channel errors have been modeled using the Gilbert-Elliot wireless channel model [9,10].

In order to validate eq. (4) we have collected statistics of throughput from two scenarios where link capacity is completely saturated (Fig. 3, scenarios 1 and 2). Namely, two nodes (scenario 1, N_{max}= 2) and four nodes (scenario2, N_{max}= 4) transmit UDP video streams to a destination node through an intermediate relay station respectively.

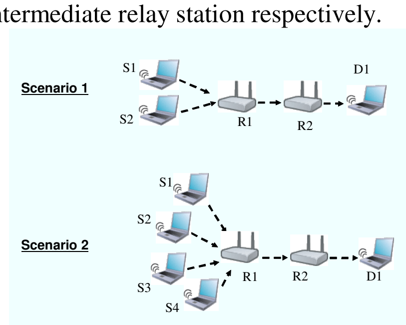

Figure 3 Multi Hop Scenarios

A. *Maximum throughput with inelastic traffic*

In Fig. 4 and Fig. 5, the theoretical results obtained with eq. (4) for many couples (L,BER) are compared with the results obtained through simulation. In both scenarios the analytical plane as for eq. (4) approximate well the simulation plane obtained through OMNET++ models. The maximum difference between the two planes is of 5% (scenario N_{max}=2, Fig. 4) and 7% (scenario N_{max}=4, Fig. 5) and a standard deviation lower than 3% for both scenarios. This provides an overall validation and assessment of eq. (4).

Eq. (4) can be used for an overall dimensioning of a multi-hop network rather then to provide accurate estimate of throughput, which in general can be hardly provided.

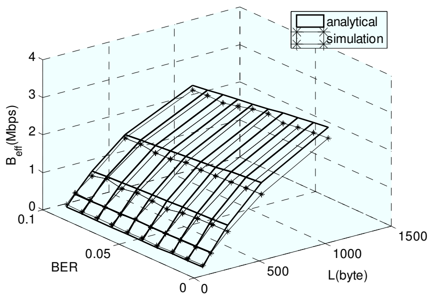

Figure 4 Multi-hop throughput assessment vs. packet length *L* and *BER* (scenario 1, N_{max}= 2)

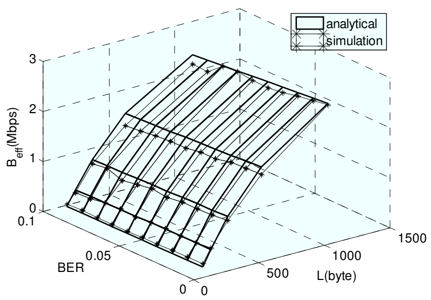

Figure 5 Multi-hop throughput assessment vs. packet length *L* and *BER* (scenario 2, N_{max}= 4)

B. Throughput in presence of elastic and inelastic traffic

It is worth highlighting that theoretical throughput calculated as in eq. (4) can be approached only when the link is saturated with sessions running on UDP, which conveys inelastic traffic. If we instead transport TCP sessions in the link, throughput can be significantly reduced on account of packet loss due to contention as well as channel impairments. TCP, unlike UDP, react to packet loss by considerably reducing the throughput. As a consequence, when bandwidth is contended between UDP and TCP sessions, UDP sessions tend to prevail over TCP sessions as their throughput is insensitive to packet loss. This behavior has been observed and commented also in the work [12].

We have then studied scenario 2 in Fig. 4 where the offered traffic is a mix of UDP and TCP sources. Namely, we have considered two UDP video streaming sources along with an HTTP and an FTP session. The BER of the wireless link has been set to 10% and packet size for all the sessions set to 1024 byte. We have then observed a reduction of the throughput from 2,3 Mbps experience with 4 UDP video sources (Fig. 6) to 1,54 Mbps when the mixed UDP and TCP sessions where used instead of 4 UDP sources. This reduction accounts for an overall loss of throughput of 33%, which affects only TCP sources, with UDP video sources maintaining a mean throughput of 580 Kbps.

This result shows that using PCF for UDP traffic is not only important to keep high-level performance for real-time traffic, generally conveyed with UDP.In order to improve performance of TCP traffic in an IEEE802.11 link, separating contention of UDP sessions with TCP sessions is also vital and this can be obtained using PCF window for UDP flows and DCF window for TCP flows.

C. End-to-end delay performance

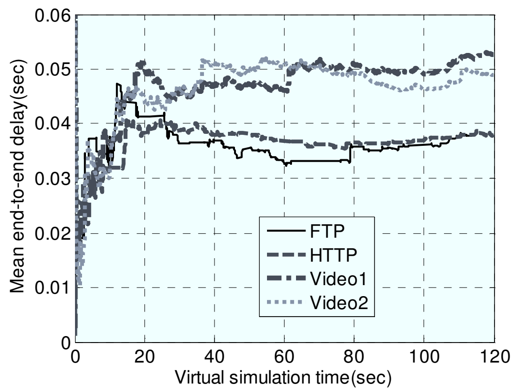

Figure 6 Multi-hop throughput assessment vs. packet length *L* and *BER* (scenario 2, N_{max}= 4)

Fig. 6 shows delay performance of the four traffic sources in scenario 2. If we compare delay statistics of the two TCP sources (FTP and HTTP flow) with the UDP ones (video1 and video2) we observe that performance of TCP transport is generally better than that of UDP transport. As no traffic control strategy is applied, delay differentiation between UDP and TCP can only depend on the interaction of UDP and TCP protocols with channel errors and with the CSMA/CA MAC. This demonstrates that a traffic control strategy is required, first of all, to invert delay performance of TCP and UDP flows, so that UDP packets are generally delivered in shorter time than TCP packets.

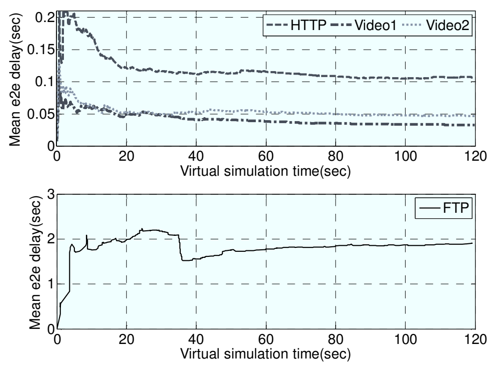

Figure 7 Multi-hop throughput assessment vs. packet length *L* and *BER* (scenario 2, N_{max}= 4)

Figure 7 shows delay statistics in the previous scenario when traffic control is used in the AP node to implement differentiated QoS management of UDP and TCP flows.

The traffic control which is used consists of two EDF schedulers, i.e. EDF1 and EDF2, and a PQ (priority queuing) scheduler [11]. EDF1 is used to schedule packets of UDP flows and EDF2 packets for TCP flows. The PQ scheduler imposes that packets can be extracted from TCP queues by EDF2 only when UDP queues are empty. In addition, UDP flows are regulated by LBs prior scheduling by EDF1. This assures that TCP traffic is not starved by excessive consumption of resources by UDP traffic, which are given absolute priority over TCP traffic by the PQ scheduler. Table I shows priority parameters used in the simulation, which sets approximate target values to be reached by the end-to-end delay of the various traffic sources.

TABLE I
EDF PRIORITY PARAMETERS

EDF1		*EDF2*	
Video1	0.040 s	HTTP	0.080 s
Video2	0.020 s	FTP	2.000 s

It can be noticed from figure 7 that the incorporated traffic control has allowed to reduce the dependency of delay performance on the IEEE802.11 MAC and invert performance of UDP and TCP flows as required. In addition, adoption of traffic control has also resulted into performance differentiation among flows transported with the same protocol. Namely, end-to-end delay for one video source has been reduced from roughly 50 ms to 35 ms. In turn, end-to-end delay for HTTP and FTP flows from initial values of 40 ms for both has been increased to 100 ms and 2 s respectively.

It is worth highlighting that the delay differentiation introduced by the traffic control has to regarded particularly good considering that the overall load of video sources corresponds to 75% of the total traffic (i.e. throughput of each video source is roughly 580 Kbps, while the total throughput of the four sources 1,54 Mbps, as discussed in subsection *A*).

4. CONCLUSION

We have developed a theoretical approach to estimate effective capacity in an IEEE802.11b link, where the medium is contended by *N* nodes and proposed an architecture for quality of service support in AP nodes. Using such traffic control strategy in relay nodes, we have showed how to invert UDP and TCP delay performance in favor of UDP traffic and determine further delay differentiation within UDP sources and TCP sources.

4. REFERENCES

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