Lessening VoIP capacity degradation in 802.11 networks with a measurement-based channel-aware scheduler *(extended abstract)*

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1. Introduction

The IEEE 802.11 DCF is known to grant all stations the same channel access probability. In ideal conditions this translates into an equal portion of the overall effective bandwidth for each user. However, as soon as some frame is not correctly received, the standard provides for the retransmission of the frame, possibly adopting more robust but less efficient modulation schemes. Such retransmissions occupy the channel at the expenses of all other stations, that consequently perceive a sensible reduction of their available bandwidth. So, the performance of the network is driven by the station with the worst link. This phenomenon, known as the *performance anomaly* of 802.11 [1], not only reduces the capacity of the network in terms of bandwidth, but also boosts up other detrimental factors, such as the jitter in frame delivery time, that may be lethal to services such as real-time voice connections.

Starting form these observations, we have designed a centralized scheduling algorithm, called *DTT* (an acronym for *Deficit Transmission Time*), whose aim is guaranteeing each downlink flow an equal time share of the channel. Most important, this goal translates into the desirable properties of proportional fairness and flow isolation [2]. The scheduler is briefly outlined in the following Section, while a thorough description can be found in [3], together with some experimental tests based on a prototype implementation with unidirectional downlink traffic. In the present manuscript we further analyze the scheduler behavior in comparison to the plain First-In First-Out (FIFO) discipline employed in commercial APs, focusing in particular on bidirectional VoIP traffic. We carried out a series of simulative tests within the framework defined by the E-model, an ITU-T standardized computational tool.

2. The proposed scheduler

A classifier, several queues and a scheduler engine are inserted between MAC and network layers. The classifier splits outgoing traffic into the queues, according to the destination MAC address. Each queue has an associated bucket that accounts for the air-time usage of the previous frames. Air time is converted into "tokens" that are used to fill/drain the buckets according to the following rules.

Once a frame transmission has been completed (either successfully or not), the scheduler computes the *Cumulative Frame Transmission Time* (*CFTT*) that includes all retransmission attempts, backoff and idle periods occurred since the frame has reached the head of the transmission queue. The bucket associated to the destination of the frame is then drained by a number of tokens equal to the CFTT. Next, this same value is divided among all non-empty queues and used to fill their buckets. At last, the next frame to be transmitted is picked from the queue whose associated bucket has the largest number of tokens. This frame is then passed to MAC layer that provides for the physical delivery.

Note that the CFTT is a deterministic measure (not an estimate, nor a prediction) of the link state: more retransmissions, possibly at lower bit rates, are carried out in the attempt to deliver the frame to stations whose channel quality is poor. Hence, stations in unfavorable positions have to wait longer before being chosen again, and the bucket with more tokens is also the one whose associated queue has transmitted less. In this way our scheduler can achieve long term fairness in air time usage and all stations are granted the same amount of downlink transmission time.

3. Performance analysis

3.1. Simulation environment

The DTT scheduler has been evaluated via simulation. The tool we have used is the OMNeT++ simulator, integrated with the Mobility Framework developed at the Technical University of Berlin, and with an 802.11b MAC layer that we have built from scratch. This tool has been validated comparing its behavior with known models. We observed negligible differences with respect to the chosen baselines (details are in [3]).

In our evaluation, all nodes are equipped with 802.11b cards, with a peak transmission rate of 11 Mbps, plus a simple automatic rate adaptation mechanism. The scheduler has been located at the AP, while all the stations always work with the plain FIFO discipline. Most nodes are close to the AP, and have a very good link to it, while one or more nodes are pretty far from the AP, thus suffering some transmission errors. Consequently they need one or more retransmissions to deliver each frame. Note that this scenario represents the typical occurrence of the performance anomaly: few nodes waste a lot of resources trying to keep up their voice connections.

All stations are involved in a bidirectional voice call in which the other end is a remote terminal connected to the AP along a wired network. Voice frames are produced by a GSM-EFR encoder and encapsulated into an IP packet, which is in turn transported by the RTP protocol. Each voice source is modeled according to the ITU-T recommendation P.59 for artificial conversational speech: the source alternates on and off periods, whose length is described by an exponential distribution with mean 1 and 1.35 seconds respectively.

3.2. The E-model

The E-model [4] is an ITU-T standardized computational method for the assessment of the quality of voice connections, as perceived by an average user. It takes into account many parameters, such as the effects of room noise, quantizing distortion, delay, and impairments due to codec and packet-loss. The primary output of the model is the scalar rating factor R, that is calculated as the sum of several factors, and ranges between 0 and 100. Since the focus of our research is on the wireless access, we monitored just two factors: the delay introduced by the wireless LAN for each packet since its arrival at the AP (T_{WLAN}), and the packet loss ratio over the 802.11 network (P_{pl}). As for the other parameters, their values have been set according to the choice of the codec and the network topology.

3.3. Simulation results

The aim of the simulations was to evaluate the number of stations allowed in the network with the users experiencing a satisfactory speech quality. In terms of the rating factor R, this means having $R = 70$. We carried out several experiments, with varying number of nodes and topologies. An exhaustive dissertation of the results is in [5], while here we just give a short report.

Fig. 1 plots the R factor when two nodes are far from the AP. The plain FIFO policy clearly does not distinguish among the stations, therefore all users experience almost the same quality of the worst station. This is because the simple FIFO-driven AP, in the effort of supplying all stations with the same bandwidth, wastes a lot of network resources. But this try has no benefit neither to the farthest users, whose connection is intrinsically hampered by propagation conditions, nor to the nearest users, whose speech quality is dragged down to the values of the worst one.

On the contrary, the adoption of DTT causes a sharp separation of the two classes of nodes. While the far users see their bandwidth shrinks, the closer ones can still perceive the channel in a good state, thus being able to sustain a high quality voice connection until the network saturates. Such a gain is the direct consequence of the principle behind the design of our scheduler, i.e. the isolation of the flows. Each user can get the maximum of its bandwidth share independently of the condition of all other stations.

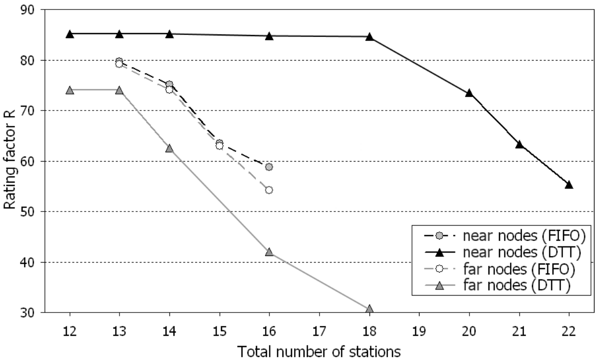

Fig. 1 – The rating factor R versus the number of stations in the network with two nodes far from the AP.

References

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