A SCHEDULING ALGORITHM FOR QoS SUPPORT IN IEEE802.11E NETWORKS

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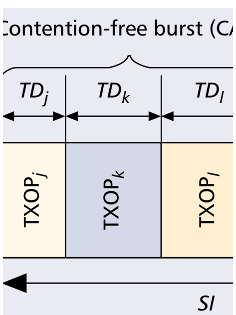

The proposed algorithm is compatible with the link-adaptation mechanisms implemented in commercial WLANs, as it limits the amount of time during which the stations control the wireless medium.

ABSTRACT

This article presents a scheduling algorithm for the IEEE 802.11e Hybrid Coordination Function under definition by the IEEE 802.11e task group. HCF can be used to provide IP quality of service guarantees in IEEE802.11e infrastructure WLANs. The Enhanced Distributed Coordination Function is mainly used for data transmission without QoS guarantees, but can also be used to decrease the transmission delay of QoS-sensitive traffic. Scheduling of queued packets follows a Delay-Earliest-Due-Date algorithm. The proposed algorithm is compatible with the link adaptation mechanisms implemented in commercial WLANs, as it limits the amount of time during which the stations control the wireless medium. The performance of the algorithm is evaluated through computer simulation and compared with the reference scheduler presented by the IEEE 802.11e task group.

INTRODUCTION

The widespread use of multimedia networking applications has created a need for quality of service (QoS) support in the Internet infrastructure. The Internet Engineering Task Force (IETF) has defined the integrated services (IntServ) and differentiated services (DiffServ) reference models for IP with QoS, which are being deployed in corporate IP networks and by Internet service providers. However, IP QoS is only possible if supported by the underlying access technology. While in wired access networks this task is facilitated by the stability of channel conditions, wireless access networks are subject to fast changes in signal-to-interference-plus-noise ratio (SINR) due to phenomena like path loss, shadowing, multipath fading, and interference. SINR, in turn, affects the bit error rate (BER) experienced by the wireless endpoints. In this environment channel capacity varies over time and space, especially when the stations are on the move. It turns out that the variability of available radio resources does not allow the network to provide hard QoS guarantees. Instead, the network must provide soft QoS guarantees constrained by a minimum channel quality.

Several scheduling algorithms were proposed for QoS provisioning in wireless networks [1]. Examples of these algorithms are those proposed in [2–4]. These algorithms consider a constant physical bit rate, which means that the transmission time for one bit is constant and the transmission time of a packet only depends on its length. They usually rely on a wireless channel model consisting of a Markov chain with two states: error-free ("good") and error-prone ("bad"). Stations that experience a bad channel are considered unable to communicate, and their resources are borrowed by other stations that experience a good channel. When the channel quality becomes good again, the stations can recover the total or a fraction of the borrowed resources. This simple theoretical model does not take into account that the BER can be lowered, for the same SINR, at the expense of physical bit rate. Reduction of physical bit rate can be achieved by selecting modulation and coding combinations that present lower bandwidth efficiency. This technique is usually designated as link adaptation or rate adaptation, and is widely used in both outdoor wireless networks and wireless LANs (WLANs.) When link adaptation is used, a station that experiences a bad channel (low SINR) may still transmit and receive albeit with a lower bit rate, which invalidates the simple Markov model. In fact, a Markov model for link adaptation would have to consider a good and a bad channel state for each available bit rate. The Automatic Rate Fallback (ARF) [5] algorithm used in WaveLAN-II products from Lucent is a simple algorithm in which transmission bit rate is adapted by the sender depending on the number of missing acknowledgment frames. Other WLAN products use similar techniques. The HIPERLAN/2 and IEEE802.11a standards for 5 GHz WLANs have almost identical orthogonal frequency-division multiplexing (OFDM)-based physical layers that specify seven and eight different bit rates, respectively [6]. These are obtained with different combinations of modulations and forward error correction (FEC) coding rates. The scheduling

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algorithms should thus be adapted to take into account link adaptation, allowing implementation in commercial WLAN equipment.

While most commercialized WLAN products nowadays are based on the IEEE 802.11 standard [7], its current medium access control (MAC) specification offers little QoS support. A special IEEE 802.11 task group (Task Group E — TGe) was created in order to specify a set of QoS enhancements to the IEEE 802.11 MAC under the designation IEEE 802.11e. These enhancements are still under discussion. The effectiveness of some IEEE 802.11e enhancements was already demonstrated [8, 9]. A recent TGe Consensus Proposal [10] provides some guidelines on how to build an IEEE 802.11e scheduler. A reference design of a simple scheduler was already presented in [11] that is compatible with the use of link adaptation. Nevertheless, this scheduling algorithm is only intended as a reference, and while it respects the minimum performance requirements [12], it is somewhat inefficient.

This article presents a proposal for a new scheduling algorithm for the IEEE 802.11e hybrid coordination function (HCF) called Scheduling Based on Estimated Transmission Times — Earliest Due Date (SETT-EDD). This is an enhancement of a previous proposal of SETT for the legacy IEEE 802.11 point coordination function (PCF), which was based on a simple round-robin scheduler [4]. The performance of SETT-EDD is evaluated through computer simulation and compared with the performance of the TGe reference scheduler.

The IEEE 802.11e MAC

The upcoming IEEE 802.11e standard defines a new operation mode for the MAC, the HCF [13, 14]. The HCF multiplexes contention-based medium access with polling-based medium access.

Contention-based medium access is designated the enhanced distributed coordination function (EDCF). IEEE 802.11 Task Group E has selected the virtual DCF (VDCF) as the EDCF mechanism to be incorporated in the upcoming IEEE 802.11e standard [15]. EDCF adds prioritization to the legacy DCF by allowing different traffic classes called *access categories*. One or more user priorities can be assigned to each access category [16]. Each user priority has a different queue and different contention window parameters. In legacy DCF, the backoff window of a station can only start after the carrier sense mechanism determines that the medium has been idle for at least a DCF interframe space (DIFS) time interval of fixed length. In EDCF, this time interval can be different for each user priority and is designated an arbitration interframe space (AIFS). An AIFS can be equal to or greater than a DIFS. Each access category contends for medium access with one carrier sense multiple access (CSMA) instance, using the backoff parameters that belong to its lowest user priority.

While EDCF prioritization is an important enhancement of legacy DCF, it is not enough to provide effective traffic protection and QoS guarantees. These can only be achieved with

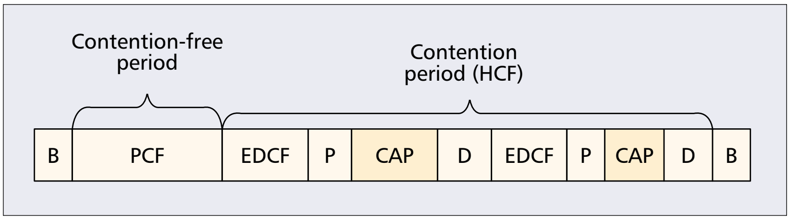


Figure 1. *Multiplexing between PCF, EDCF, and CAPs. PIFS and DIFS intervals are denoted Pand D, respectively; beacon frames are denoted B.*

polling-based medium access and centralized scheduling. The polling mechanism of HCF is similar to the legacy PCF. It is controlled by the hybrid coordinator (HC), which is typically located at the access point (AP) in infrastructure WLANs. A new set of frames is defined. These remain similar to the legacy PCF frames, but have a QoS control field added in order to identify the traffic class and other QoS-related attributes: QoS NULL, QoS DATA, QoS CF-ACK, QoS CF-POLL, QoS DATA+CF-ACK, QoS CF-ACK+CF-POLL, QoS DATA+CF_POLL, and QoS DATA+CF-ACK+CF-POLL.

In HCF, the HC is allowed to start contention-free bursts (controlled access periods, CAPs) at any time during the contention period after the medium remains idle for at least a PCF interframe space (PIFS) interval, shorter than DIFS (Fig. 1). A contention-free burst ends after the medium remains idle for a DIFS interval. The HCF contention-free mechanism is more flexible than legacy PCF because the latter has a fixed length and must occur periodically after a beacon frame. IEEE 802.11e stations are still allowed to support PCF besides HCF [17].

A contention-free burst is formed by a sequence of transmission opportunities (TXOPs). A TXOP is a period of time in which a station or the HC can transmit a burst of data frames separated by a short interframe space (SIFS) interval, which is the shortest interframe space. A TXOP is always initiated by the HC. The HC starts an uplink TXOP issuing a poll request to a station. A downlink TXOP is started by the HC with the transmission of a data frame. A TXOP ends when at least one of the following conditions is met:

- Transmission of a data frame with the nonfinal (NF) flag set to 0, which means that there are no further frames queued for transmission.
- Expiration of the TXOP duration, which is given by default by the variable **dot11DefaultCPTXOPLimit**. It can also be explicitly set by the HC in beacons, association response frames, or poll request frames.
- A polled station allows the wireless medium to remain idle for a PIFS interval in an uplink TXOP.

The proportions of contention-based and polling-based transmission in HCF is specified, as contention-based access is still needed for important management tasks (e.g., association of stations with the AP during handover). The rate and proportion of contention-free bursts are given by the variables **dot11CAPRate** and **dot11CAPMax**, respectively. The **dot11CAPRate** variable specifies the fraction of time that can be

The aggregate Maximum Burst Size is equal to the sum of the Maximum Burst Size of all the TSPECs reserved by the station. An aggregate token bucket is initialized with the aggregate Maximum Burst Size. Tokens are added to the token bucket at the aggregate Mean Data Rate.

used for contention-free bursts, expressed in units of microseconds per 64 µs (e.g., a **dot11CAPRate** value of 32 means that at most half the time can be spent in contention-free bursts). On the other hand, the variable **dot11CAPMax** specifies the maximum duration of a contention-free burst. Together, the two HCF variables define a time token bucket whose state is given by the CAP timer. The CAP timer is initialized to zero and counts upward at a rate defined by **dot11CAPRate** until it reaches the maximum value of **dot11CAPMa**x. At any time, the HC can deduct from the CAP timer a number of units equal to or less than its current value and start a contention-free burst whose duration in microseconds corresponds to the number of deducted units.

In order to be included in the polling table of the HC, a station must issue a QoS reservation by means of special QoS management action frames [18]. A separate reservation must be made for each traffic stream (TS), which is a unidirectional stream of high-ayer packets (MAC service data units, MSDUs) requiring QoS guarantees. A TS is described by a traffic specification (TSPEC), whose main parameters are:

- **Mean data rate** (ρ)**:** average bit rate for transfer of the packets, in units of bits per second.
- **Delay bound (D):** maximum delay allowed to transport a packet across the wireless interface (including queuing delay), in milliseconds.
- **Nominal MSDU size (L):** nominal size of the packets, in octets.
- **User priority (UP):** priority to be used for the transport of packets in cases where relative prioritization is required (e.g., it can be used for admission control). It is based on IEEE 802.1d priority levels [19], which go from 0 (lowest) to 7 (highest).
- **Maximum MSDU size (M):** maximum size of the packets, in octets.
- **Maximum Burst Size (MBS):** maximum size of the data burst that can be transmitted at the peak data rate, in octets
- **Minimum PHY rate (R):** physical bit rate assumed by the scheduler for transmit time and admission control calculations, in units of bits per second.
- **Peak data rate (PR):** maximum bit rate allowed for transfer of the packets, in units of bits per second.

These parameters are very similar to those included in the IP FlowSpecs defined in RFC 1363 [20] and used by IntServ and DiffServ. This facilitates QoS mapping between the IP and MAC layers. The first three parameters are more important, and in [11] it is suggested that they should be mandatory.

In order to make a TS reservation, the station issues an Add_TS_QoS Action frame to the HC, carrying the respective TSPEC. The Del_TS_QoS action frame finishes a reservation.

The HC issues polls to stations and not to individual TSs (i.e., the HC schedules TS aggregates). According to [10] the HC uses the individual TSPECs of a station to calculate an aggregate service schedule, which includes the following main parameters:

Minimum TXOP duration (*mTD*): minimum TXOP duration allocated to the station. The *mTD* is equal to the maximum packet transmission time for any of the station's TSPECs. The maximum packet transmission time of a TSPEC reservation *i* is the time required to send a packet of size *M* at the minimum PHY rate (packet fragmentation is omitted for sake of simplicity),

$$mTD_i = MAX\left(\frac{M_i}{R_i}\right). \qquad (1)$$

Maximum TXOP duration (*MTD*): maximum TXOP duration allocated to the station. The *MTD* is bounded by the transmission time of the aggregate maximum burst size (see below).

Minimum service interval (mSI): minimum time between the start of successive TXOPs allocated to the station, in units of microseconds. Given a service interval for each TSPEC (calculated as L/ρ), the *mSI* contained in the service schedule is equal to the smallest service interval for any TSPEC that belongs to that station.

Maximum service enterval (*MSI*): maximum time allowed between the start of successive TXOPs allocated to the station, in units of microseconds. Although no guidelines are provided in [10] to calculate this parameter, in this article it is assumed that *MSI* is related to the lowest delay bound (*D*) among the station's TSPECs, being calculated as:

$$MSI = \beta \cdot (D - MTD) \qquad (2)$$

with $0 < \beta \leq 1$.

The HC may use an aggregate token bucket specification to police a station's admitted flows. It must derive the aggregate mean data rate and aggregate maximum burst size to establish the aggregate token bucket specification. The aggregate mean data rate is equal to the sum of the mean data rates of all the TSPECs reserved by the station. The aggregate maximum burst size is equal to the sum of the maximum burst size of all the TSPECs reserved by the station. An aggregate token bucket is initialized with the aggregate maximum burst size. Tokens are added to the token bucket at the aggregate mean data rate.

A reference scheduler is proposed in [11], and is designated in this article as the TGe scheduler. It only uses the mandatory TSPEC parameters. Two parameters need to be derived in this simple scheduler:

Service interval (*SI*): interval between TXOPs, which is the same for all stations. According to the scheduler specification, it should be a submultiple of the beacon frame interval. It should also be less than or equal to the lowest *MSI* of all stations. Consequently, when a new reservation is accepted, *SI* must be recalculated if the resulting *MSI* drops below its current value.

TXOP duration (*TD*): fixed duration of a TXOP, which is calculated for each station. For a station *j* that has made *n* TSPEC reservations, TD_j can be calculated according to the following expression:

$$TD_j = \sum_{i=l}^{n} MAX(NTD_i + O, mTD_i + O), \qquad (3)$$

where O is the overhead due to PHY and MAC headers, IFSs, acknowledgment frames, and poll frames. NTD_i is the nominal TXOP duration for TSPEC reservation i and can be calculated as follows (packet fragmentation is omitted for sake of simplicity):

$$NTD_i = \frac{\left\lceil \frac{SI \times \rho_i}{L_i} \right\rceil \times L_i}{R_i}. \tag{4}$$

In this simple scheduler, TXOPs of different stations are put together, forming contention-free bursts with period SI and duration equal to the sum of all TXOPs (Fig. 2).

Stations that experience a good channel (high SINR) are able to transmit at a bit rate higher than R_i. In this case packet transmission takes less time than expected for the reservation. These stations could in this case try to transmit data at an average data rate higher than the negotiated ρ_i. As already said, a token bucket mechanism can be used to police this overload traffic. Nevertheless, nothing should prevent the station from using the saved time for retransmission of lost frames (e.g., due to link adaptation delay) when needed. When the saved time is not needed, it can be used to anticipate the TXOPs of other stations, transmit multicast/broadcast frames, or resume EDCF contention.

When a station experiences low SINR, it may have to transmit or receive at a physical bit rate lower than the minimum PHY rate (R). In this case, packet transmission takes more time than it should. The TGe scheduler ensures that the traffic of other stations is protected and their QoS not jeopardized by limiting the total transmission time of each station j to TD_j in each SI.

Admission control is trivial with the TGe scheduler. The total fraction of transmission time reserved for contention-free transmission (CAP reservation, CR) for m stations at any given moment can be easily calculated as follows:

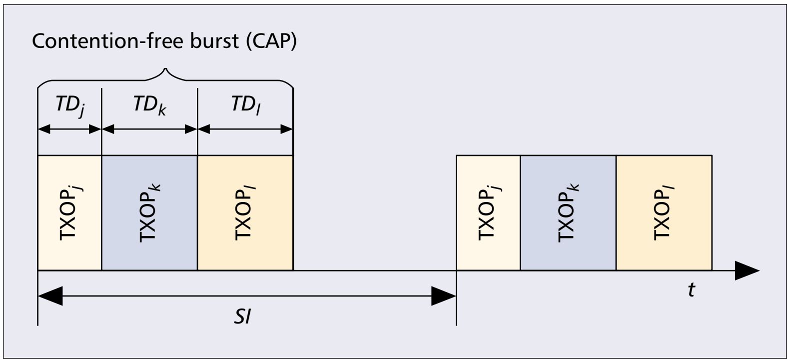
$$CR = \sum_{j=1}^{m} \frac{TD_j}{SI}. \tag{5}$$

In order to check if a new reservation can be accepted, the HC only needs to check if the new reservation k plus the current CR is lower than or equal to the maximum fraction of time that can be spent by contention-free bursts (the normalized CAP rate):

$$\frac{TD_k}{SI} + CR \le \frac{dot11CAPRate}{64\mu s}. \tag{6}$$

Scheduling Based on Estimated Transmission Times: Earliest Due Date

While the implementation of the TGe scheduler is easy, it can be quite inefficient in some scenarios. The TD of a station is always the same and corresponds to the transmission time of an M-sized packet or a burst of average size (whichever takes longer) at the minimum PHY rate. While this may be suitable for traffic types that present small bursts of constant size (e.g., voice over IP, VoIP), some traffic types like MPEG-4 video present bursts of variable size formed by several packets (e.g., an MPEG-4 I-frame is usually much larger than a P-frame or B-frame). With the TGe scheduler, transmission of a long burst in several TD-sized TXOPs, spaced by SI periods, can lead to significant transmission delay. It would be convenient to have more flexibility regarding the size of the station's TXOPs, while enforcing TD as the average TXOP duration. This can be achieved with a simple technique similar to the CAP timer used to limit polling-based transmission in HCF: a token bucket of time units or TXOP timer. The TXOP timer of station j increases at a constant rate equal to TD_j/mSI_j, which corresponds to the total fraction of time the station can spend in polled TXOPs. The TXOP timer has a maximum value equal to MTD_j. The time spent by a station in a polled TXOP is deducted from the TXOP timer at the end of the TXOP. The station can be polled only when the value of the TXOP timer is greater than or equal to mTD_j, which ensures the transmission of at least one packet at the minimum PHY rate.



■ Figure 2. *A temporal diagram of the TGe scheduler with reservations made by three stations* j, k, *and* l. *The TXOPs of all stations are transmitted in sequence, forming contention-free bursts with period* SI.

To poll all the stations with the same period (equal to SI) may not be adequate because some traffic sources generate bursts sporadically, while others generate bursts more frequently. In our proposed scheduler, each station has an independent service interval equal to mSI. If the due time to poll a station is t, the next poll shall be issued on a time t' that satisfies the relation

$$t + mSI \le t' \le t + MSI. \tag{7}$$

Time instant $t + mSI$ is the instant after which the next poll can be done, equivalent to the release time in the real-time scheduling theory. Time instant $t + MSI$ is the maximum time by which the next poll has to be done, or deadline time.

The HC's scheduler has to decide which station to poll first, among those that satisfy Eq. 7 at a given moment. As the stations are scheduled to transmit over a common communication channel, one at each time instant, this type of scheduling problem is similar to a common and simple scenario of the real-time scheduling theory, (i.e., scheduling of asynchronous periodic

TSPEC	VoIP (G.729A)	Video (MPEG-4)
Mean data rate	24 kb/s	630 kb/s
Delay bound	60 ms	60 ms
Nominal MSDU size	60 octets	1024 octets
Maximum MSDU size	60 octets	1024 octets
Maximum burst size	120 octets	14,894 octets
Peak data rate	24 kb/s	1.5 Mb/s
User priority	6	5
Minimum PHY rate	24 Mb/s	24 Mb/s

Table 1. *TSPECs for VoIP and video.*

tasks with periods less than or equal to their deadlines on a single server). Real-time scheduling theory has already settled that a discipline which issues polls by nondecreasing deadlines — Earliest Deadline First (EDF) or Earliest Due Date (EDD) — is optimal in a wide set of real-time scheduling problems, including the one that we have just described [21]. Because of its optimality and simplicity, we have decided to use it with SETT. More specifically, we have decided to use Delay-EDD [22], which is a variant of EDD that calculates deadlines based on delay bounds. As already mentioned, the deadline for the start of a TXOP in SETT-EDD is $t + MSI$. The admission control calculations for SETT-EDD remain similar to those for the TGe reference scheduler, and obey Eqs. 5 and 6, in which SI is now replaced with mSI_j for each station j (recall that each station can have a different service interval in SETT-EDD).

Finally, a short note on the hardware/firmware implementation of SETT-EDD. Finding the station that is ready for transmission and whose deadline is the lowest is an operation of complexity $O(n)$, where n is the number of stations. A solution that can substantially reduce the computational complexity is one based on a circular memory. This circular memory is a vector of c elements, where each element represents a time slot. As the real-time clock increases, the elements of the vector are scanned in increasing order, wrapping around to the first element of the vector, after the last element of the vector is scanned. Each element of the vector points to the stations that can be polled after the instant corresponding to the beginning of the slot (i.e., the stations that satisfy Eq. 7). However, the time interval corresponding to each time slot of the vector must be shorter than the lowest mSI of all the stations, and the time interval corresponding to the full vector dimension must be longer than the highest MSI. Each time a station is polled, it goes again to some position of the vector, corresponding to the time instant of the poll plus its mSI. The scheduler has therefore to analyze a small number of stations each time, because the stations' poll times are spread along the vector, as they are typically not synchronized. The references to the stations whose poll times satisfy Eq. 7 can be inserted on a separate list of schedulable stations, among whom the scheduler uses the Delay-EDD discipline. These list elements are deleted once the respective stations finish their TXOPs.

Simulation Parameters and Traffic Source Models

The simulations consider three types of traffic sources: bursty data (e.g., HTTP sessions), VoIP, and video.

The model for bursty data sources is Source Type 1 defined for performance evaluation of 802.14 [23]. It consists of a Poisson arrival distribution. This model generates the following message sizes (in octets) and respective probabilities: (64, 0.6), (128, 0.06), (256, 0.04), (512, 0.02), (1024, 0.25), and (1518, 0.03). Each bursty data source generates a data rate of 200 kb/s.

The audio source model generates 60-octet messages periodically with an interval of 20 ms, resulting in a bit rate of 24 kb/s. This is a suitable model for G.729A [24] with the RTP/UDP/IP overhead and without voice activity detection.

For the video source model we have used a trace of a real MPEG-4 video stream of an e-learning session (Lecture Room-Cam video stream [25]).

VoIP and data sessions are bidirectional (i.e., each station is the source of an uplink flow and the sink of a downlink flow for the session it runs) while video sessions are unidirectional downstream sessions. Bursty data is transmitted with EDCF only, and is assigned the lowest priority (0). TSPEC reservations for VoIP and video are listed in Table 1. A higher priority was assigned to VoIP as telephony services are more time-critical, having more stringent delay and jitter constraints than non-real-time video streaming. It should be noted that we calculate two separate service schedules for each station, corresponding to the uplink and downlink directions. For the MSI calculation based on the delay bound given by Eq. 2, we consider $\beta = 33$ percent.

A **dot11CAPRate** value of 21 µs means that, on average, only one third of each unit of 64 µs can be used in contention-free bursts. This means that the maximum fraction of time occupied with polling-based transmission is approximately 33 percent. This is the limit for TSPEC reservations above which Eq. 6 becomes false. The maximum duration of a contention-free burst is configured as 8 ms. The CAP Timer update interval is configured as 5120 µs. This follows the recommendation expressed in [17] that the CAP timer should be updated at uniform intervals that are multiples of 64 µs, and no less than 1024 µs (Table 2).

In this model, packet losses happen when the maximum allowed retransmission attempts (**aShortRetryLimit** and **aLongRetryLimit**) are exceeded or the maximum queuing delays of packets exceed the configured limit (**dot11MSDULifetime**).

The IEEE 802.11a PHY layer was selected for the simulations. The adopted IEEE 802.11a error model is the same as presented in [26]. According to this model, the original received frame is the first frame that arrives with enough received signal power (C) to allow recognition of the physical layer preamble. Reception of other frames that overlap the time span of the original frame contribute to the cumulative co-channel interference power level (Σ^I). Taking into account the background noise power level (N), the SINR of the original reception is calculated as $C/(\Sigma^{I+N})$. A simplification is made that the SINR is constant throughout the time span of the original frame. Packet error probability is then calculated as a function of SINR according to the framework presented in [27]. The latter assumes binary convolutional coding and hard-decision Viterbi decoding with independent errors at the decoder input. This framework only allows calculation of an upper bound for the packet error rate because exact calculation is impossible when the only available information is the average bit error probability. This is due to the fact that the bit error pattern at the output of a Viterbi decoder is typically bursty, even if errors are independent at the input.

The simulation model also considers link adaptation. An adaptation of Automatic Rate Fallback (ARF) to IEEE 802.11a is used [28]. When no acknowledgment is received after two consecutive data frame transmission attempts, the sender decreases the bit rate. Each additional missing acknowledgment causes the bit rate to decrease. If 10 consecutive data transmissions are successful, the sender tries to increase the bit rate. This can also happen if a timeout (60 ms) expires during which no data transmission failure occurs.

In some simulations it is useful to abstract link adaptation in order to isolate the performance of the scheduler. In this case an ideal SINR-based link adaptation will be used in which it is assumed that the sender knows in advance the SINR experienced by the receiver for any frame transmission. Before sending a frame, the sender selects the transmission mode that maximizes throughput for the respective SINR value. Figure 3 depicts the maximum throughput as a function of SINR for 1024-octet packets according to the model presented in [27].

IEEE 802.11e MAC	aSlotTime	20 µs
	Beacon interval	100 ms
	aFragmentationThreshold	1024 octets
	aRTSThreshold	500 octets
	SIFS	20 µs
	PIFS	40 µs
	DIFS	60 µs
	dot11MSDULifetime (VoIP+video)	60 ms
	dot11MSDULifeTime (data)	200 ms
	aShortRetryLimit aLongRetryLimit	7
	dot11DefaultCPTXOPLimit	3000 µs
	dot11CAPRate	21 µs
	dot11CAPMax	8000 µs
	CAP timer update time	5120 µs

Table 2. *MAC parameters.*

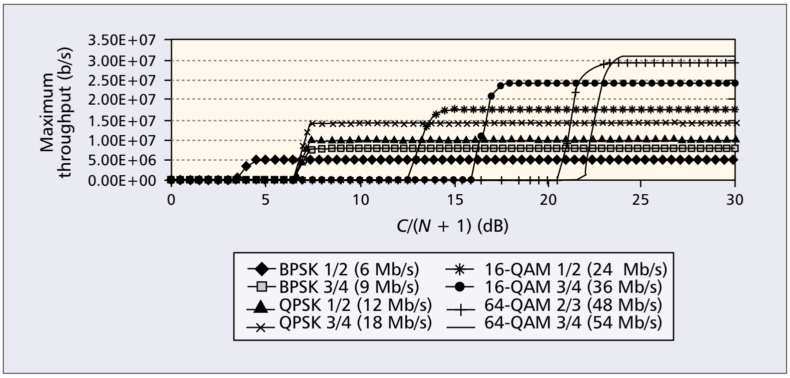


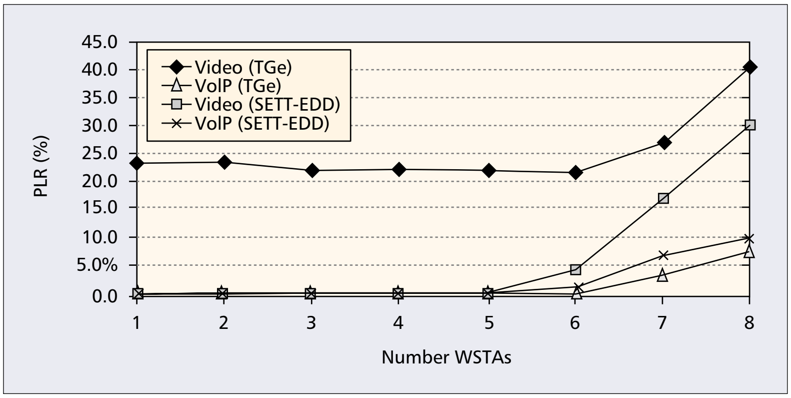
Figure 3. *Maximum throughput as a function of SINR for 1024-octet packets in IEEE 802.11a.*

Simulation Results

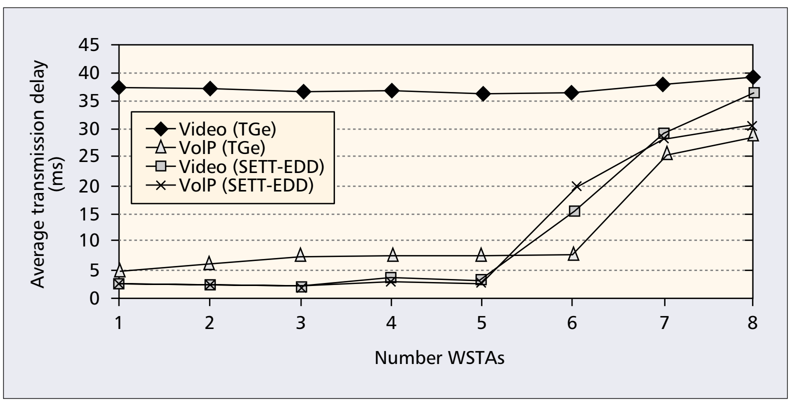
The first simulation scenario (S1) allows a direct performance comparison between the TGe scheduler and SETT-EDD. It considers an increasing number of stations forming a star around an AP, all located at a distance of 20 m from it. Each station runs a video session simultaneous with a G.729A VoIP and a bursty data session. The reservation limit is exceeded when six or more stations are associated with the AP, *CR* then being greater than the fraction of time allocated to polled transmissions (33 percent). Ideal SINR-based link adaptation is assumed. At the distance of 20 m the optimal transmission bit rate between each station and the AP is 24 Mb/s, the worst case that still respects the minimum PHY rate defined in the TSPECs. VoIP and video are not transmitted with EDCF, so the measured performance is only related to the scheduler. The packet loss ratio (PLR) and average transmission delay of video and VoIP are depicted in Figs. 4 and 5 respectively.

As predicted, the video PLR is always higher than 20 percent for the TGe scheduler, while in SETT it is negligible for less than six stations. This is also reflected in the average transmission delay. As already said, this result is due to the bursty nature of video traffic, whose variable burst size is incompatible with the fixed TXOP duration of the TGe reference scheduler.

Below the reservation limit of six stations, VoIP transmission delay is slightly lower in SETT-EDD than in the TGe scheduler. The reason is the same as for video. Although bursts are smaller for VoIP than for video, the fixed TXOP duration of the TGe scheduler limits the number of transmissions per service interval, which can cause delay accumulation due to retransmissions. For more than six stations, the PLR and transmission delay of VoIP become slightly lower in the TGe scheduler, which could lead us to think the TGe scheduler is more efficient for VoIP services in some situations. But this phenomenon is simply due to the fact that video packet losses,



■ Figure 4. *Packet loss ratio of video and VoIP traffic (scenario S1).*



■ Figure 5. *S1: Average transmission delay of video and VoIP traffic (scenario S1).*

always higher than 20 percent, free some of the available TXOP time for use by VoIP.

Although not depicted in the graphics, with six or more stations, VoIP performance degradation is due to the uplink direction only. This can easily be explained by the adopted implementation of both scheduling algorithms. By having independent service schedules for each direction and each station, and due to the fact that video sessions are downstream only, the downstream service schedules result from the aggregation of VoIP and video TSPECs, while the uplink service schedules are only based on a VoIP TSPEC. As VoIP is assigned higher priority than video, it is able to seize downlink TXOP time that otherwise would be used for video transmission.

Finally, it should be mentioned that data traffic experienced low transmission delays and negligible losses as two thirds of the total time was assigned to EDCF. Simulations were done in which video and VoIP were transmitted during the EDCF as well, which resulted in very low transmission delays and packet losses, as already shown in [8]. Nevertheless, EDCF is not subject to accurate admission control, which means that it offers no guarantees when the capacity of the WLAN becomes saturated. In these conditions, the stations that are granted TSPEC reservations keep their QoS levels, while the others degrade their performance.

The purpose of the second scenario (S2) is to show the impact of movement and link adaptation on the scheduler's performance. The first simulation (S2a) has a station moving in a straight line with a speed of 1 m/s between two positions located respectively at points (–10 m, 15 m) and (10 m, 15 m) considering the AP's location at the origin (0, 0). The station runs a bidirectional VoIP application only. Rayleigh multipath fading is applied based on the Jake's model implementation presented in [28]. The second simulation (S2b) is identical to the first except that the station moves from point (–10 m, 10 m) to point (10 m, 10 m) and hence remains closer to the AP. ARF link adaptation is tested against ideal SINR-based link adaptation and fixed bit rate at 24 Mb/s (the minimum PHY rate in the TSPECs). The latter could in fact be approximated by a two-state Markov chain. As can be concluded from Fig. 3, the transition between bad and good states for 24 Mb/s is only of 2 dB (between 12.5 dB and 14.5 dB).

Results for simulation S2a are listed in Table 3. ARF adapts very slowly to the rapid changes due to multipath fading. It only decreases the bit rate each time a data frame is lost, taking 10 successful transmissions or 60 ms to increase the bit rate again. In S2a, the retransmissions required by ARF to reach the optimal bit rate exceed the TXOP time assigned to the station in SETT-EDD, causing transmission to lag and 8.2 percent of the packets to be lost due to queuing delay (vs. the expected 2.5 percent with ideal link adaptation). ARF is still much better than fixed bit rate, which presents 20.4 percent of packet loss. This is because transmission at 24 Mb/s is not possible for SINR values below 12.5 dB (bad state), which occur in deep fading situations. Performance of the TGe scheduler is slightly lower than SETT-EDD in all configurations due to the reasons already mentioned in the first scenario, the flexibility offered by the TXOP timer in SETT-EDD.

In simulation S2b, the WSTA is closer to the AP; thus, the optimal bit rate is usually higher than 24 Mb/s. Results for this simulation are listed in Table 4. TSPEC reservations are seldom exceeded, which translates into lower packet loss and delay in all configurations. It also explains why the PLR is lower for fixed bit rate (1.1 percent for SETT-EDD and 1.3 percent for the TGe scheduler) than ARF (2.0 percent for SETT-EDD and 2.5 percent for the TGe scheduler). This inefficiency of ARF is due to failed attempts to raise the physical bit rate. Ideal link adaptation continues to present better performance than ARF and fixed bit rate. Performance of the TGe scheduler is again slightly lower than SETT-EDD.

Conclusions

This article presents a scheduling algorithm named Scheduling Based on Estimated Transmission Times — Earliest Due Date (SETT-EDD), which can be used for QoS provisioning in IEEE 802.11e WLANs. The performance of SETT-EDD was evaluated through computer simulation and compared with the performance of the reference scheduler proposed by the IEEE 802.11e task group. The simulation scenarios considered the integration between IEEE 802.11e and IEEE 802.11a in an infrastructure WLAN. The results demonstrate that SETT-EDD achieves better performance than the TGe scheduler, especially in the transmission of streamed video.

The impact of link adaptation on the performance of scheduling algorithms is also demonstrated. It is shown that the Markov chain with two states is not accurate in modeling the status of the wireless channel in WLAN technologies with link adaptation like IEEE 802.11a and HIPERLAN/2. These results also motivate research on more efficient SINR-based link adaptation techniques and their integration with the MAC and power control mechanisms of existent WLAN technologies.

Link adaptation algorithm	Packet loss ratio		Average transmission delay	
	SETT-EDD	Tge	SETT-EDD	TGe
ARF	8.2%	8.7%	46 ms	46 ms
Ideal SINR-based link adaptation	2.5%	2.7%	35 ms	38 ms
Fixed bit rate = 24 Mb/s	20.4%	23.3%	40 ms	47 ms

■ **Table 3.** *The impact of link adaptation on the scheduler's performance (simulation S2a).*

Link adaptation algorithm	Packet loss ratio		Average transmission delay	
	SETT-EDD	TGge	SETT-EDD	TGe
ARF	2.0%	2.5%	38 ms	39 ms
Ideal SINR-based link adaptation	0.4%	0.8%	25 ms	37 ms
Fixed bit rate = 24 Mb/s	1.1%	1.3%	34 ms	43 ms

■ **Table 4.** *The impact of link adaptation on the scheduler's performance (simulation S2b).*

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