Packet loss-ratio based scheduler: an adaptive scheduling scheme to increase the number of multimedia applications in WLAN

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Abstract: The reference scheduler in IEEE 802.11e fails to maintain the goal of quality of service (QoS) support and high network efficiency simultaneously. In this paper we investigate channel access schemes to increase the number of multimedia applications in wireless local area network (WLAN) while maintaining the QoS requirements. We have developed a scheduling scheme, named as packet loss-ratio based scheduler (PLS) that takes into account the packet losses occurred while allotting the transmission time to a multimedia stream. To implement this scheme the access point (AP) requires only the queue size information and can be implemented within the format of IEEE 802.11e. The simulation results show that the PLS can accommodate more number of traffic streams and can also randomise the packet loss from a traffic stream, which is a requirement for multimedia coder-decoders (CODECs) to conceal the distortion.

Keywords: wireless local area networks; WLAN; HCF controlled channel access; HCCA; packet loss; quality of service; QoS; multimedia; IEEE 802.11e.

Reference to this paper should be made as follows: Jibukumar, M.G., Datta, R. and Biswa, P.K. (2011) 'Packet loss-ratio based scheduler: an adaptive scheduling scheme to increase the number of multimedia applications in WLAN', Int. J. Ultra Wideband Communications and Systems, Vol. 2, No. 1, pp.34–43.

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1 Introduction

The significant benefit of carrying multimedia applications in a wireless local area networks (WLAN) is that it can eliminate the need of a dedicated network for multimedia services. Providing multimedia applications such as voice, streaming audio/video, network gaming and so on in WLAN is challenging because of its real-time requirements and unpredictable traffic characteristics. The distributed coordination function (DCF) in IEEE 802.11 has failed to meet the increasing demand of users for throughput-intensive and delay-sensitive multimedia applications. In DCF the packets of multimedia applications and delay insensitive data packets are treated in the same way.

The IEEE 802.11e, an amendment of IEEE 802.11 to support multimedia applications, provides a relative channel access priority scheme called enhanced distributed channel access (EDCA) or a contention free channel access called HCF controlled channel access (HCCA) to the packets of multimedia applications to meet their quality of service (QoS) requirements (IEEE 802.11e Part 11, 2005). In EDCA the access priority to a traffic stream (TS) is realised by changing the inter-frame spaces (IFS) and contention window size. Thus EDCA increases the number of idle slots and number of packet collisions which results in the decline of network throughput. The studies carried out by Shankar et al. (2004) proved that network throughput obtained in EDCA is less than that in DCF. This is because of the aggressive nature of EDCA. In this context several papers (Ye et al., 2007; Minyoung and Namhyun, 2006; Narbutt and Davis, 2007; Scalia et al., 2006) have pointed out that, to optimise the system performance with network load variations, tuning of EDCA parameters are required. The goal of EDCA is to provide service guarantees. It is shown that EDCA can provide real time service to highest priority TSs at the price of lower priority TSs, which starves especially at high network load (Engelstad and Østerbo, 2005; Lindgren et al., 2003). Many multimedia applications demand small packet delay and periodic access of the channel. However, EDCA provides only a relative service differentiation among TSs (Ni et al., 2004; Bianchi and Tinnirello, 2003).

It is well know that only contention free channel access can meet absolute QoS requirements of multimedia. The IEEE 802.11e provides a reference scheduler and an admission controller algorithm for the implementation of contention free channel access. In the next subsection we give an overview of the contention free channel access in IEEE 802.11e, called as HCCA.

1.1 HCF controlled channel access

In HCCA, a QoS station (QSTA) negotiates with the access point (AP) using traffic specifications (TSPEC) to get admission to the polling list. A TSPEC specifies the QoS requirements required for the TS. The important parameters in the TSPEC are nominal packet size (S), mean data rate (λ), minimum physical data rate (R), delay bound (D), maximum packet size (M) and maximum service interval (max SI). The maximum time interval between two successive polls for the TS is called Max SI and if it is not specified, it is taken as the delay bound. Once TSPEC negotiation is successful, the TS is admitted and the AP has to offer transmission opportunity (TXOP) to meet the QoS requirements.

The scheduler first calculates the value of scheduled service interval (SI), the time interval used by the AP to poll QSTAs. The SI is calculated as a duration that is smaller than the max SI of all admitted TSs and a sub-multiple of the beacon interval. This selection criterion of the SI is to provide the QoS requirement of the lowest-delay constrained TSs. Thus, any admitted TS can be served within its maximum SI specification. The scheduler then calculates the transmission opportunity duration (TXOP) that has to be allotted to each TS in a scheduled SI.

The calculation of transmission opportunity duration needed for a TS in a scheduled SI is based on the average number of packets that may be generated during that period. The transmission opportunity duration for the jth admitted TS (TXOP) is calculated as follows:

\[
TXOP_j = \max \left( \frac{S_j}{R} + X \right) \times G_j \times \frac{M_j}{R} + X
\]

(1)

where \( G_j \) is the number of packets that arrived at mean data rate (\( \lambda_j \)) during a scheduled SI in the jth TS and calculated as,

\[
G_j = \left[ \frac{SI \times \lambda_j}{S_j} \right]
\]

(1a)

In equation (1), X is the packet transmission overhead time that includes IFS, acknowledgement (ACK) frame transmit time and polling frame transmit time and other variables are the QoS specifications given in the TSPEC of jth admitted TS. Similar to a round robin scheduler, the AP next allots transmission opportunity duration to each of the admitted TSs (Figure 1). HCCA reference scheduler uses mean values of the multimedia traffic given in the TSPEC for calculating the TXOP and thus fails to adapt the VBR and self similar characteristic of multimedia application (Ansel et al., 2006; Boggia et al 2007; Rashid et al., 2008). We now
briefly discuss multimedia traffic characteristics, some proposed reference schedulers and the motivation behind our work.

**Figure 1** IEEE 802.11e super frame structure

1.2 Multimedia traffic characteristics and some proposed schedulers

Out of several multimedia applications, video transmission shows an increasing demand and its QoS support in WLAN is challenging due to its characteristics. Several research works have shown that most of the coder-decoders (CODECs) for video generate self-similar traffic (Adas and Mukherjee, 1995; Sheluhin et al., 2007). Self-similarity implies the presence of a long-range dependence (LRD) in the traffic rate. That is, there exist non-negligible positive correlations in the burst behaviour of the traffic over many time scales ranging from milliseconds to seconds to minutes or more. The self-similarity in network traffic raises serious concerns for network traffic management, particularly for video-on-demand (VOD) service providers and backbone network providers. Bursts will exist across many time scales and positive correlations in traffic will adversely affect the QoS provided to the network users (Duffield, 1995; Elwalid et al., 1995). The only way to provide reasonable QoS is to operate the network at a sufficiently low utilisation so that queuing effects, buffer overflows, and network congestions are negligible. However, this results in inefficient use of the transmission facilities. Fortunately, there are several published works that show statistical multiplexing gain is still possible with self-similar traffic sources (Krunz et al., 1995; Krunz and Tripathi, 1996). That is, even though the aggregated traffic remains self-similar, the relative variability of the aggregated stream is low. This can be effectively utilised in a contention based channel access. But the scheduler used by IEEE 802.11e, called as ‘reference scheduler’ failed to utilise this.

Several studies have shown that this reference scheduler is effective only for constant bit rate (CBR) services and fails to achieve the QoS requirements of multimedia applications and network efficiency simultaneously. The variable bit rate (VBR) of multimedia traffic and its small packet delay requirements are accounted for this failure. Many multimedia applications generate VBR traffic. Feedback based scheduling schemes (Ansel et al., 2006; Boggia et al., 2007; Grilo et al., 2003; Rashid et al., 2008) which use queue size information from stations, can adapt the VBR characteristic of multimedia TSs. In IEEE 802.11e, a QSTA can communicate its queue size to the AP via a field in the MAC header. The AP can utilise this information for proper scheduling the transmission of packet without violating their QoS requirements.

The ‘FHCF scheduler’ proposed by Ansel et al. (2006) uses an estimation of queue lengths without taking into account the urgencies for service of the packets in the admitted stream queues. Rashid et al. (2008) proposed a scheduler called ‘pro HCCA scheduler’ using earliest dead-line first (EDF) algorithm with traffic prediction to reduce the packet delay. The EDF scheduling algorithms can absorb bursts of small duration but fails to support if it prolongs for a long time (i.e., if the traffic shows self-similar properties) during network congestion. These studies do not incorporate the small packet loss that is allowed for the multimedia applications to optimise the number of supported multimedia applications while keeping the QoS requirements. A proportional integral (PI) controller for meeting the target packet delay in HCCA channel access mechanism has been proposed by Boggia et al. (2007). Here the packet dropping due to delay limit violation is not considered.

Noh et al. (2007) proposed a scheduling strategy to meet the user-level QoS guarantee in audio video transmission. This scheduling scheme is similar to HCCA reference scheduler where the TXOP assignment is determined by mean data rate and a user defined parameter $\alpha$. By varying the parameter $\alpha$ different user QoSs can be attained. In Kuo (2008) proposed a measurement-based dynamic transmission opportunity scheme, which adaptively allocates resources to a VBR video on the basis of the estimation of future traffic demand to support efficient QoS transmission of VBR video. The selectivity function scheduler (Bourawy et al., 2009) differentiates the number of packets into the packets remaining in the queue from previous TXOP, and new packets arrived during the scheduled $\delta f$. A station knows its TS queue size at the beginning of each polling phase and estimate the number of packets that arrive to find the queue length at the end of the TXOP. However, all these schedulers are optimised to reduce the packet delay and do not include the other characteristics of multimedia applications such as packet loss ratio, random error distribution etc.

A scheduler based on the queue size information from the QSTA can utilise the statistical multiplexing gain of the video streams and can accommodate more number of TSs as compared to the reference scheduler. The rest of the paper is organised as follows. In Section 2 we present our proposed scheduling scheme named packet loss-ratio based scheduler (PLS) and its implementation aspects in IEEE 802.11e networks. We present the simulation results obtained by using this scheduler in a WLAN and its comparison with reference scheduler in Section 3. Section 4 concludes the paper with a brief discussion.

2 Packet loss-ratio based scheduler

Here we present a scheduling scheme with an objective to maximise the number of multimedia TSs in a WLAN. We
focus on the scheduling strategy that is to be followed during the limited resource conditions. The pro HCCA scheduler design (Rashid et al., 2007) does not consider the scheduling strategy that need to be followed when network congestion occurs. Further, the inaccuracies in the predicted traffic leads to wrong calculation of contention free period (CFP) and affects the energy minimisation schemes used in IEEE 802.11 WLAN. This prompts us not to include any type of traffic predictors in our scheme. Instead, in our proposed scheme, a node sends the queue size information to the AP using the QoS fields in the packet format of IEEE 802.11e and does not consume any additional channel resources. It also guarantees a data transmission opportunity to a node frequently so that the AP can easily capture the traffic pattern of packet generation.

2.1 Basic principle of PLS

Even though video TSs show small delay constraints, they can afford some packet losses. If the packet losses occur within the specified limit at random intervals, most of the video CODEC can conceal the distortion occurred without affecting the mean opinion score (MOS) (Hassal, 2000). Our proposed scheduler named as PLS is efficient when network congestion occurs. The scheduler drops packets from TSs based on its recent packet losses. The PLS work similar to the EDF scheduling scheme when there is no network congestion. We presume that the AP know about the packet loss-ratio permitted for a multimedia TS.

In each scheduled SI the PLS allot a minimum transmission time (\( \sigma \)) to a QSTA. This time is required to send its queue size to the AP if its buffer is reported empty in the previous scheduled SI. The actual time allotted to a TS in a scheduled SI depends on various factors such as network load, packet loss occurred, its queue size etc. Next we describe the calculation of scheduled SI in the PLS.

Scheduled SI

The first step in PLS scheme is to select a suitable value for the scheduled SI that meets the QoS requirements of all admitted TSs. Here afterwards we refer this value as \( (SI_{PLS}) \). In PLS, the calculation of scheduled SI is slightly different from the existing schemes (Boggia et al., 2007; IEEE 802.11e Part 11, 2005; Rashid et al., 2008). Here the transmission time to a TS is based on the queue size/backlogged packets in a QSTA and as such there is an unavoidable delay occurring in this process due to the delay in communicating the queue size to the AP. This delay can be up to one scheduled SI. As PLS uses static scheduling for CBR TSs, the SI for them can be up to the delay bound of CBR TSs. We have selected one third of the delay bound of VBR TSs as its SI. This selection criterion of SI is to provide a freedom of at least two SI to the AP for serving the reported backlogged packets.

The scheduled SI \( (SI_{PLS}) \) is calculated as a duration that is smaller than the SI of all admitted TSs and a sub-multiple of the beacon interval. For an illustration, let us consider five TSs with packet delay bounds of \( (50, 200, 250, 150, 300) \) ms. Let us assume that the first TS is a CBR application. In this case, the SI of the TSs are given by \( \{50, 65, 82, 50, 100\} \) ms. If the beacon interval used by the AP is 400 ms, then 50 ms or a lower value \( \{40, 25, 20\} \) ms can be selected as the scheduled SI \( (SI_{PLS}) \).

Transmission time to a TS in a SI_{PLS}

The AP allows transmission time to admit TSs in a fair and efficient manner. For doing this, accurate estimation of the network load in the next scheduled SI is essential. The estimation of network load in the next scheduled SI is done as follows.

The AP maintains a register \( (Z) \) to keep the traffic generation patterns of TSs in the previous \( (D_{-1}) \) scheduled SI where

\[
D_{j} = \left[ \frac{\text{Delay bound of } j\text{th traffic stream}}{SI_{PLS}} \right]
\]  

The \( (i, j) \)th element of the register \( (Z_{ij}) \) indicate the backlogged packets (in bytes) of \( j\)th TS that have already experienced a queuing delay of \( i \) scheduled SI. Thus, the first row corresponds to the entire backlogged packet in the buffer of each TS and the second row excludes the packets that have experienced a delay of one scheduled SI. Therefore the last element of a column \( (i.e., (D_{j} - 1) \)th element) indicates the packets that have delay limit in the next scheduled SI.

As an illustration, (refer Figure 2) let us consider four TSs with delay bounds \( \{200, 250, 300, 300\} \) ms. Let us assume that the AP uses a scheduled SI of 50 ms. From the figure it can be seen that the fourth TS has 80 backlogged bytes and 45 bytes were generated in the pervious scheduled SI. Also, it has 5 bytes in its buffer that have delay limit in next scheduled SI. Similarly the second stream has no packet that has delay limit in the next scheduled SI. Therefore using such a register, the AP can easily estimate the network load and transmission priority in the next scheduled SI. As shown in Figure 2, the AP calculates the total offered network load in the next scheduled SI as 1330 bytes. Out of these 1330 bytes, 35 bytes in the TSs \( \{1,3,4\} \) have delay limit in the in the next scheduled SI. Therefore the minimum transmission time required to avoid packet loss in the next scheduled SI (\( TXOP_{min-PLS} \)) is the summation of the time for transmitting packets that have delay limit in the next scheduled SI and the mandatory minimum transmission opportunity duration \( (\sigma) \) that has been discussed earlier in this section. Mathematically it can be written as,

\[
TXOP_{min-AP} = \sum_{j} C_{j}
\]  

where \( C_{j} \) is given by,
if \( Z_{D_{ij}} \neq 0 \)
\[ C_j = \text{TXOP}_{\text{min},j} \]
else
\[ C_j = \sigma \]

Figure 2 Register \((Z)\) of backlogged traffic in QSTAs kept in the AP

The number of packets and its generation time

Here \( \text{TXOP}_{\text{min},j} \) is the transmission time required to transmit the packets of the \( j\)th TS that have delay limit in the next \textit{scheduled SI}. It can be calculated as follows

\[
\text{TXOP}_{\text{min},j} = \frac{Z_{D_{ij}} S_j}{X + \frac{S_j}{R}} \quad (4)
\]

In equation (4) the first term on the right hand side corresponds to the number of packets that have delay limit in the next \textit{SI} and second term corresponds to the transmission time of a packet of \( j\)th TS.

The total transmission time required to transmit the reported backlogged packets in the next \textit{scheduled SI} \((\text{TXOP}_{\text{TN},\text{PLS}})\) can be written as,

\[
\text{TXOP}_{\text{TN},\text{PLS}} = \sum_j \text{TXOP}_j \quad (5)
\]

where \( \text{TXOP}_j \) is given by,

if \( (Z_{ij} \neq 0) \)
\[
\text{TXOP}_j = \left[ \frac{Z_{ij}}{S_j} \right] \times \left( X + \frac{S_j}{R} \right) \quad (5.a)
\]
else
\[ \text{TXOP}_j = \sigma \]

In equation (5.a), \( \left[ \frac{Z_{ij}}{S_j} \right] \) is the average number of packets in the next \( SI \) for the \( j\)th TS and second term corresponds to the transmission time of a packet of \( j\)th TS. The \( \text{TXOP}_{\text{TN},\text{PLS}} \) in the above example is the time required to transmit 1,330 bytes.

Figure 3 Transmission sequence in PLS

In our scheme, the transmission time for multimedia packets is limited to \( \alpha \times \text{SI}_{\text{PLS}} \) where the term \( \alpha \) is a constant \((\alpha < 1)\) selected by the AP (Figure 3). The remaining time of the \textit{scheduled SI} is used for transmitting best effort traffic using the contention based channel access. Depending on the network load and delay limit of the backlogged packets, there can be three different cases. The AP calculates the transmission time to the admitted TSs in the next \textit{scheduled SI} for the three cases as follows:

Case 1  \( \alpha \times \text{SI}_{\text{PLS}} \) is less than \( \text{TXOP}_{\text{TN},\text{PLS}} \): The AP allots time to all TSs that are sufficient to deplete its buffer. In such case all the entries in the register are set to zero. The transmission time (TXOP) for the \( j\)th TS is calculated as follows:

\[
\text{TXOP}_j = \left[ \frac{Z_{ij}}{S_j} \right] \times \left( X + \frac{S_j}{R} \right) \quad (6.a)
\]

\[ Z_{ij} = 0 \text{ for all } i \text{ and } j. \]

Case 2  \( \alpha \times \text{SI}_{\text{PLS}} \) is lower than \( \text{TXOP}_{\text{TN},\text{PLS}} \) and greater than \( \text{TXOP}_{\text{min},\text{PLS}} \): The AP initially allots time to transmit the packets with delay limit in the next \textit{scheduled SI}. The remaining time \((\alpha \times \text{SI}_{\text{PLS}} - \text{TXOP}_{\text{min},\text{PLS}})\) is allotted to all TSs based on their urgencies (EDF first scheduling policy). If the transmission time (TXOP) allotted to \( j\)th TS is for transmitting \( A_j \) bytes, then TXOP\(_j\) is given by:

\[
\text{TXOP}_j = \left[ \frac{A_j}{S_j} \right] \times \left( X + \frac{S_j}{R} \right) \quad (6.b)
\]

In equation (5.a), \( \left[ \frac{A_j}{S_j} \right] \) is the average number of packets allowed to transmit from for the \( j\)th TS in the next \( SI \) and second term corresponds to the transmission time of a packet of \( j\)th TS.
Packet loss-ratio based scheduler

The register (Z) is modified as follows

\[
\text{if } (Z_{ij} - A_j) > 0 \\
Z_{ij} = Z_{ij} - A_j \\
\text{else} \\
Z_{ij} = 0
\]

(6.c)

Case III: $\alpha \times Sl_{PLS}$ is less than TXOPminPLS: The AP cannot allot sufficient time to transmit the packets that have delay limit in the next scheduled SI. In this case, the AP has to adjust the transmission time of a stream by allowing some packet loss. The allotted transmission time (TXOP) to a TS in this case is based on the packet loss-ratio register (PLR) maintained by the AP. This procedure will be explained later in this section. The transmission time (TXOP) allotted to a TS to transmit $A_j$ bytes in this case also is calculated as in equation (6.b). The list is updated by deducting the last entry $\left(\left(Z_{D,-1,j}\right)_{th}\right)$ from all corresponding higher entries i.e., $Z_{ij} = Z_{ij} - Z_{D,-1,j}$.

Once the transmission time for each TS (for the three cases as above) has been calculated, the AP shifts each row of the register downwards by one unit. In the next scheduled SI the AP allots the transmission time to the TSs like a round robin scheduler. A QSTA communicates the queue size to the AP by using the QoS control field specified in the data packet. Thus at the end of each scheduled SI the AP knows about the queue size of each TSs and this is entered in the first row of the register.

For downlink TSs, the AP finds out the number of packets arrived for each TS in it and updates the register (Z) for backlogged packets (PLR) and calculates TXOP in a way similar to uplink traffic scheduling. In the case of downlink transmission, the AP can use the actual packet size for the calculation of packet transmission time instead of average packet size in the case of uplink packet transmission. Since the AP has priority in channel access and has knowledge about accurate multimedia traffic patterns, the resource allocation is less complex than uplink and there are several modes to transmit downlink packets.

2.2 Packet loss-ratio register

In our proposed scheme the AP maintains a register, known as PLR, to track the packet loss experienced by the TSs. An entry in the packet loss-ratio register (PLR) gives the numbers of future scheduled SI that must elapse for dropping a packet from the stream without affecting MOS. It is related to the loss-ratio (LR) allowed for the TS and the instant at which the previous packet loss has occurred. For example, Figure 4 indicates that the first TS has recently suffered some packet losses and therefore any further packets drop before next 15 scheduled SI will affect MOS. In this scenario, if network congestion occurs in the next scheduled SI the AP allow to drop some packets from the second or the fifth TSs.

Figure 4 Packet loss-ratio register

Thus, if network congestion occurs, the AP reduces the time allotted to the TS that has smallest value in the PLR by an amount of time sufficient enough to drop one packet. Subsequently the entry in the PLR is updated as follows: If a packet is dropped from $j$th TS, its entry in PLR is updated to

\[
PLR_j = PLR_j + \frac{1}{LR_j \times G_j}
\]

(7)

where $LR_j$ and $G_j$ [equation (1.a)] are the loss-ratio allowed and average number of packets that arrived in one scheduled SI in the $j$th TS respectively. This process continues until the total TXOP in the next scheduled SI falls below $\alpha \times Sl_{PLS}$. If there is no packet loss for a stream in a scheduled SI the value in the PLR register corresponding to the TS is reduced by one. We limit the value of PLR to

\[
\frac{5}{Sl_{PLS}(in\ s)}
\]

This is because of the real time nature of the multimedia traffic where in the effect of packet loss is not accumulated.

3 Simulation results

Our simulations were performed in MATLAB. We used the physical layer specifications of IEEE 802.11a. The physical and MAC layer parameters used in our simulations are summarised in Table 1. In both scheduling schemes, a QSTA employ EDF procedure for transmitting its packets in the allotted TXOP. The AP responds to each packet transmission with an ACK frame. We did not consider the packet error during the transmission.

The simulation studies use both CBR and VBR multimedia TSs. The VoIP applications are modelled as CBR TSs at 64Kbps and generate 160 bytes in every 20 ms (Shankar et al., 2004). To generate video TSs we used the video traffic traces of movies (VFST, 2007). Two types of video traffic traces are used for evaluating the performance of the schedulers: interactive and non interactive. The interactive video traffic uses H.263 coded frame traces and non-interactive video traffic uses MPEG-4 coded frame traces. We have selected the video traffic traces for one hour duration (Star Wars IV, Jurassic Park I and Silence of the Lambs) and large frame sizes are fragmented into
packets of small size. We use RTP/UDP/IP header of 40 bytes. We have simulated all the network scenarios for 400 s. A QSTA having video TS generates packets by randomly selecting a 400 s duration form the one hour duration. Table 2 provides the QoS requirements of the TS. In reference scheduling, the entire contention free duration is allotted to the admitted TSs based on their data rate except for CBR TSs. The CBR TSs use static scheduling as given in IEEE 802.11e.

Table 1  MAC and physical layer parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>RTP + UDP + IP header</td>
<td>40 bytes</td>
</tr>
<tr>
<td>MAC header</td>
<td>36 bytes</td>
</tr>
<tr>
<td>ACK size</td>
<td>14 bytes</td>
</tr>
<tr>
<td>Beacon interval</td>
<td>600 ms</td>
</tr>
<tr>
<td>DIFS</td>
<td>34 μs</td>
</tr>
<tr>
<td>σ</td>
<td>32 μs</td>
</tr>
<tr>
<td>Physical header (PLCP)</td>
<td>20 μs</td>
</tr>
<tr>
<td>SIFS</td>
<td>16 μs</td>
</tr>
<tr>
<td>Physical data rate</td>
<td>24 Mbps</td>
</tr>
<tr>
<td>Basic data rate</td>
<td>6 Mbps</td>
</tr>
</tbody>
</table>

Table 2  Multimedia traffic QoS specifications.

<table>
<thead>
<tr>
<th>Application</th>
<th>VoIP</th>
<th>Interactive video</th>
<th>Non-inter. video</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mean packet size (bytes)</td>
<td>160</td>
<td>750</td>
<td>850</td>
</tr>
<tr>
<td>Max. packet size (bytes)</td>
<td>-</td>
<td>1,000</td>
<td>1,000</td>
</tr>
<tr>
<td>Mean data rate</td>
<td>64 kbps</td>
<td>400 kbps</td>
<td>800 kbps</td>
</tr>
<tr>
<td>Delay bound</td>
<td>60 ms</td>
<td>200 ms</td>
<td>400 ms</td>
</tr>
<tr>
<td>Loss-ratio</td>
<td>0%</td>
<td>3%</td>
<td>3%</td>
</tr>
</tbody>
</table>

Average number of QoS failure

In addition to the conventional performance measures to analyse the simulation results, we introduce a new performance measurement parameter to indicate time dependency of the loss characteristics. Here, the simulation time is divided into non-overlapping intervals of small durations (W). The packet loss-ratio of each TS is calculated in these intervals. The packet loss-ratio of jth stream in the ith interval is given by

\[ LR_{ij} = \frac{N_{LOSS \_w_j} \times SI_{PLS}}{G_j \times W} \]  

(8)

where, \( N_{LOSS \_w_j} \) is the number of packets dropped from the jth stream in the ith interval and \( G_j \) [equation (1.a)] is the average number of packets arrived in \( SI_{PLS} \). This is compared with the loss-ratio limit of the TS. If the \( LR_{ij} \) exceeds the loss-ratio limit, it is considered as failure for meeting the QoS, i.e.,

\[ \text{if} \ (LR_{ij} > LR) \]
\[ n_j = 1 \]
\[ \text{else} \]
\[ n_j = 0 \]

The average number of QoS failure (ANF) during the simulation time is calculated as follows

\[ \text{ANF} = \frac{1}{N} \sum_{j=1}^{N} \sum_{i=0}^{M-1} n_{ij} \]  

(9)

where M is the number of intervals used for averaging and N is the number of admitted TSs.

Simulation results and discussions

We have chosen the scheduled SI (SI_{PLS}) as 30 ms that satisfies the conditions described in Section 2. We have limited the CFP in scheduled SI to 20 ms. Figure 5 shows the cumulative distribution function (CDF) of the packet-delay of the non-interactive video TSs at different network loads. In reference-scheduling when the network load is small, most of the packets were transmitted with very small delay. For example, it is seen that for a total network load of 4 Mbps (three non-interactive video TSs, three interactive video TSs and 6 VoIP TSs) 85% of the packets are transmitted with a delay less than 40 ms. But as the network load is increased to 6 Mbps (five non-interactive video TSs, seven interactive video and 3 VoIP), most of packets are found to experience delay above 250 ms. At low network load, the allotted TXOP to a stream is high compared to what is actually required. But as the network load increases, the allotted time to the TS reduces proportionally.

Figure 5  CDF of the packet delay (see online version for colours)

Note: Non-interactive video

In PLS the packet delay is small even at 6 Mbps. When the network load is increased to 7 Mbps (seven non-interactive video TSs, eight interactive video TSs and 3 VoIP) the packet delay is also increased. Even in this case 70% of packets experience delays less than 80 ms. It is seen that
most of the packets experience a delay above 30 m. It is due to the unavoidable delay experienced in communicating the queue size to the AP which we have discussed in Section 2. Figure 6 shows the CDF of the packet delay of interactive video. This also shows similar characteristics as non-interactive video stream.

**Figure 6** CDF of the packet delay (see online version for colours)

![CDF of the packet delay](image)

Note: Interactive video

Figures 7 and 8 show the total required time (TXOPTN_PLS), total time allotted to all TSs, total used time by the TSs and packets dropped in the scheduled SI in a typical simulation. In reference scheduling it can be seen that TXOPTN_AP switches to a high value at random. This is due to the self-similar characteristic shown by the video TSs. By these characteristics, if multimedia TSs are not properly served, the backlogged packets will increase to a high value. But this type of behaviour is not visible in PLS. Here the scheduler allot transmission time based on the backlogged queue size and can use multiplexing gain to oppose self-similar characteristics.

**Figure 7** Performance of PLS at 6.5 Mbps (see online version for colours)

![Performance of PLS at 6.5 Mbps](image)

Note: Offered load consist of 5 VoIPs, five interactive video TSs and five non-interactive video TSs.

There is a large difference between allotted time and time used in reference scheduling. This is because of the fact that for some TSs the allotted transmission time is more than sufficient to transmit its packets. This will not affect the system efficiency directly because unutilised time can be used by best effort traffic. However this will result in inaccurate calculation of contention free duration and affect the energy saving schemes used in WLAN. In PLS the AP calculates contention free duration accurately and it can give instructions to other non-QSTAs to remain idle during this period to save their energy.

In PLS the packets are dropped when the total required time (TXOPTN_PLS) is higher than $\alpha \times S_{PLS}$ for few subsequent scheduled SI. In reference scheduling, packets are dropped even if the total allotted time is higher than the total required time (TXOPTN_PLS). This is due to the inappropriate allocation of transmission time among QSTAs. Therefore, static scheduling fails to capture the VBR characteristics of multimedia TSs.

Finally, for finding the serving capacity of the schedulers we have increased the number of VBR video TSs and measure the QoS. The Tables 3 and 4 show the number of TSs and QoS obtained. It is seen that the PLS can support ten non-interactive applications as compared to eight in reference scheduling. In case of interactive video, the PLS support 20 non-interactive applications whereas 14 in the case of reference scheduling. This shows that the PLS scheduler can perform better compared to reference scheduler.

Figures 9 and 10 show the failure-occurrences during the simulation interval of a typical interactive TS. Here the packet loss-ratio is measured for every 2 sec. If this exceeds the allowed loss-ratio limit of the stream, it is considered as a failure of scheduler. It can be seen that in reference scheduler the packet loss occurs in burst and exits for long duration. However, in PLS the packet loss is rare and occurs
randomly. This is another salient feature of the PLS scheduler.

Figure 9  Timing of the QoS failure of a TS using reference scheduler

Note: Number of interactive streams: 15.

Figure 10  Timing of the QoS failure of a TS in PLS

Note: Number of interactive streams: 21

In the simulations of scheduling (HCCA and PLS) schemes, we use identical time for contention and contention free channel access (max 20 msec). Normally, in HCCA to meet the QoS requirements of multimedia applications excess transmission time is allocated than what is actually required (Figure 8). However, PLS allocates the transmission time based on the queue size. The contention free time can be calculated as done earlier (refer Figure 7) and hence PLS can easily implement the power saving schemes proposed in IEEE 802.11e. The performance improvement of PLS scheduler is due to the dynamic allocation of the CFD among admitted TSs and do not affect the best effort traffic in WLAN.

Next, we compared the performance of the PLS with proHCCA scheduler (Rashid et al., 2007). The proHCCA scheduler does not consider the packet loss during the TXOP allocation. In Figure 11 we show the simulation studies with regard to proHCCA scheduler. The figure shows that for the same QoS, the number of TSs that can be admitted in proHCCA scheduler is slightly less than that of PLS (∼18–19 interactive video streams) (refer Figure 10 for PLS). Figure 11 also shows that during network congestion, proHCCA scheduling strategy results in bursty packet losses as compared to the rare and random packet losses in PLS. This desirable performance of PLS over proHCCA scheduler is due to the weightage given to the recent packet losses during the TXOP allocation.

Figure 11  Timing of the QoS failure of a TS in proHCCA scheduler

Note: Number of interactive streams: 19

In this paper we focus on a scheduling strategy that increases the number of admitted streams while maintain the QoS requirements during the network congestion. The network congestion that causes packet dropping (due to the packet delay limit) may be either due to the increased packet generation rate or due to the variable wireless channel quality. Since the AP allots the TXOP to a TS and can determine the number of successful packet transmissions in the allotted TXOP, it (AP) can easily identify the number of dropped packets from a TS irrespective of the cause of the packet drop. Thus the PLS can be implemented in any network scenario.

4 Conclusions

In this paper, we proposed a scheduler named packet loss-ratio scheduler for WLAN to increase the number of multimedia application WLAN. The implementation of packet loss-ratio scheduler requires the packet generation patterns of the TS and packet loss occurred in the previous scheduled SI. To estimate these parameters in our scheme...
the AP requires only the queue size information from the QSTAs in each scheduled SI. The packet loss-ratio based scheduler works similar to the EDF scheduling scheme when there is sufficient time to schedule the reported queue size. When network congestion occurs, the scheduler drops packets from the TSs based on its recent packet losses. The simulation results show that the proposed scheduler can accommodate more number of TSs as compared with reference scheduler given in 802.11e. In addition to that the proposed scheduler can randomise the packet loss from a stream which is a requirement for multimedia CODEC to conceal the distortion.

References