A Novel Scheduler Design for Wireless Video over 802.11aa Networks Using Priority Weighting and Dropping

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Abstract

In the newly released 802.11aa-2012 protocol, intra access category prioritization (IACP) and drop eligibility indicator (DEI) enable stream classification service (SCS) with graceful degradation for robust audio and video streaming. IACP introduces the need for scheduling between the primary and alternative AC queues (AC_VI and AAC_VI) for differentiating real-time and non-real-time video streams. This paper proposes a novel cross-layer design for the scheduler between AC_VI and AAC_VI, which combines a real-time video importance scheme in the Application layer and a priority weighting and dropping algorithm (PWD) in the MAC layer, where priority weighting is applied only to AC_VI and priority dropping to both AC_VI and AAC_VI. The results show that the proposed design outperforms the conventional ones, including IACP-RR, ICAP-WRR, and SCS-WRR, with substantial performance gains for both real-time and non-real-time video streams via AC_VI and AAC_VI. Such a win-win game, not possibly achieved by the conventional designs, shows the true power of PWD.

Keywords: *802.11aa; intra access category prioritization; stream classification service; graceful degradation; priority scheduling; priority dropping*

1. Introduction

During the past decade, video streaming has become the main source for a wide spectrum of video applications in the Internet, including *non-real-time* and *real-time* applications, such as IPTV and video conferencing respectively. However, high quality video streaming over wireless networks has been a challenging issue because wireless channels are unreliable, error-prone in signals, and time-varying in bandwidth.

With the dominance of IEEE 802.11 technologies in wireless LANs, there has been a strong demand for quality video delivery over 802.11 wireless LANs. However, the original version of *media access control* (MAC) function, known as *distributed coordinated function* (DCF) [1], could only support a *best-effort* (BE) service and thus cannot meet the QoS requirement for video steaming. This is due to the fact that DCF adopts only a single queue and thus provides no service differentiation among different traffic types.

To provide *quality of service* (QoS) for multimedia applications, the second version of MAC function for the scenario of channel contention, called *enhanced distributed channel access* (EDCA), was thus proposed in the 802.11e protocol [2, 3]. EDCA is based on the concepts of *multiple access category* (AC) *queues* and *virtual collision handling*. In other words, EDCA adopts *four* AC queues of descending medium access

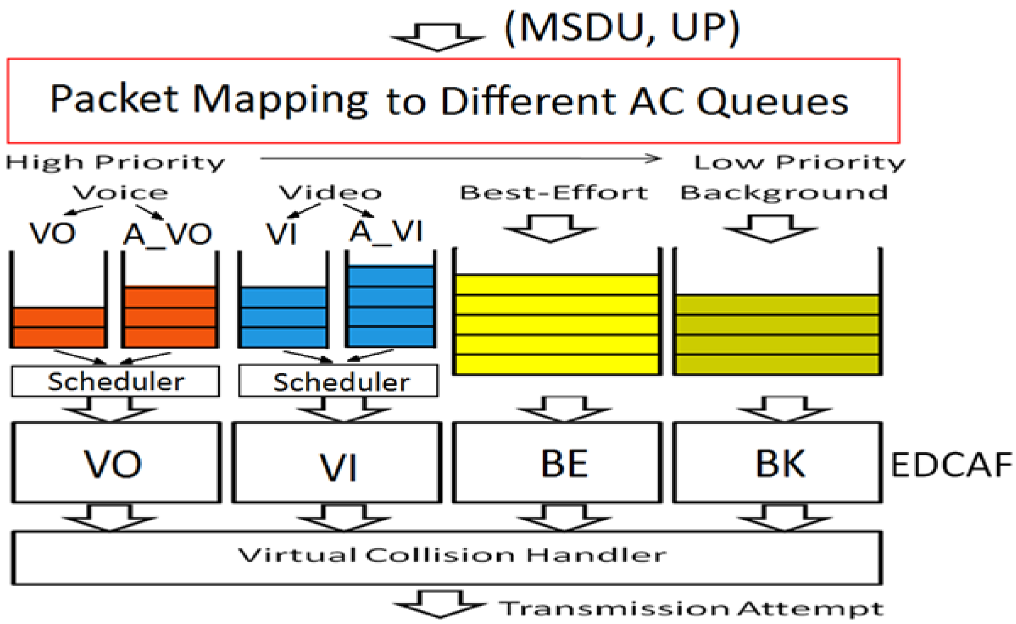

Figure 1. IACP and SCS in 802.11aa

resources to differentiate the *voice* (AC_VO, or AC[3])) and *video* (AC_VI, or AC[2]) traffic types from others, such as *best-effort* (AC_BE, or AC[1]) and *background* (AC_BK, or AC[0]). The transmission priority of each AC queue is characteristic of a set of resource parameters, including *arbitrary inter frame space* (AIFS) for *fixed backoff*, *contention windows* (CW_{min} and CW_{max}) for *random backoff*, and *transmission opportunity limit* (TXOPlimit) allowing for *multiple packet transmissions* within the allocated time limit of each obtained medium access. On the average, an AC with *shorter backoff times* and a *larger transmission opportunity* has a higher packet transmission probability. The *in-station collisions* among these ACs are resolved by the *virtual collision handler*. Despite the service differentiation of video from other traffic types, there is still a fundamental problem with EDCA which limits its performance: no further service differentiation among the video packets or flows is possible since the AC_VI queue is still FIFO-based (*passive*) and thus still suffers from the *full-queue* and *lock-out* effect.

In the literature, there have been some studies to tackle this problem [4-8]. Based on video packet importance and prioritization, one of these studies *statically* mapped video packets into different ACs without changing their FIFO feature [4], while the others [5-8] adopted some sort of *active queue management* (AQM) algorithm to *dynamically* map those video packets of *less importance* into *lower-priority* ACs than AC_VI, such as AC_BE or AC_BK, according to the congestion level of AC_VI. Although all of these studies could achieve some performance gains in video transmission quality, they must also face a common problem, *i.e.*, *traffic impact* to the *lower-priority* ACs due to those downward mapped video packets.

Targeting at MAC enhancements for *robust audio and video streaming*, a new IEEE standard called 802.11aa was released at the end of May in 2012 [9]. 802.11aa is a much more evolved version of 802.11e, covering several advanced issues, such as *groupcast with reties* (GCR), *stream classification service* (SCS), *overlapping basic* *service sets* (OBSS), and *interworking with* 802.1AVB [10]. Among these, SCS can be viewed as an evolved version of the 802.11e EDCA mechanism.

SCS provides two new QoS services: (1) *intra AC prioritization* (IACP), *i.e.*, differentiation between separate video streams that originally belong to the same AC, and (2) *graceful video quality degradation* in bandwidth shortage. As seen in Figure 1, the IACP service can be achieved by introducing the concept of *alternative* AC (AAC). In other words, the *single-queue video* access category evolves to a *double-queue*s structure. Hence, service differentiation between *real-time* and *non-real-time* video streams is now possible by mapping them to AC_VI (*primary queue*) and AAC_VI (*secondary queue*) respectively. Similarly for *voice*. On the other hand, *graceful video quality degradation* can be realized by activating the *drop eligibility indicator* (DEI) bit to sacrifice lower-priority video streams in bandwidth shortage. By confining video streams within the *primary* and *alternative* queues and assuming the same EDCAF function for the double-queues structure of AC_VI, SCS can obviously avoid the aforementioned *traffic impact* problem to AC_BE or AC_BK. However, the *double-queues* structure of AC_VI introduces the need for a scheduler before passing video packets to the EDCAF function, as seen in Figure 1. The design of such a scheduler is implementation-specific, and can profoundly affect the video transmission performance.

This paper proposes a novel cross-layer design called *prioritized weighting and dropping* (PWD). For comparison, we have also implemented two conventional schedulers for IACP such as *round-robin* (IACP-RR) and *weighted round-robin* (IACP-WRR) [11], and a variant of WRR for SCS using the DEI bit (SCS-WRR). Our experimental results show that the proposed PWD design outperforms the conventional ones.

The rest of this paper is organized as follows. Section 2 describes the design principles and technical details of PWD. The simulation environment is described and the results are discussed and analyzed in Section 3. Section 4 concludes this paper.

2. Prioritized Weighting and Dropping (PWD)

The objective of our proposed PWD design is to implement an advanced scheduler which can not only differentiate *real-time* and *non-real-time* video streams, but also achieve a *win-win game* for *both* of them in terms of further enhanced video transmission qualities than the aforementioned conventional designs. Obviously, this is challenging because of a fundamental problem: given the same EDCAF (*i.e.*, the same channel access resources) in front of the *primary* and *alternative* queues under the constraint of *zero impact* to other non-video traffic flows, it is intrinsically difficult for the conventional schedulers like RR and WRR to achieve *fairness* and *priority* in such a *double-queues* structure *simultaneously*. In other words, RR (based on *equal weighting*) achieves *fairness* but no *priority*; on the other hand, WRR (based on *unequal weighting*) emphasizes the *priority* of AC_VI by sacrificing the *fairness* of AAC_VI.

For service differentiation between *real-time* and *non-real-time* video streams, it is important to understand that their QoS constraints are quite different: *non-real-time* streams are only sensitive to *packet loss* which occurs in the case of *full queue* (*i.e.*, heavy congestions), whereas *real-time* streams are both *loss*- and *delay*-sensitive. Thus, the *real-time* QoS constraint is more stringent, and *long queue delay* (*i.e.*, *effective packet loss*) can also degrade the *real-time* video transmission quality, in addition to *packet loss* during transmission. It is thus obvious that AC_VI deserves a larger weighting factor than AAC_VI. Namely, WRR is a better scheduler than RR. Unfortunately, a larger weighting factor for AC_VI also means less

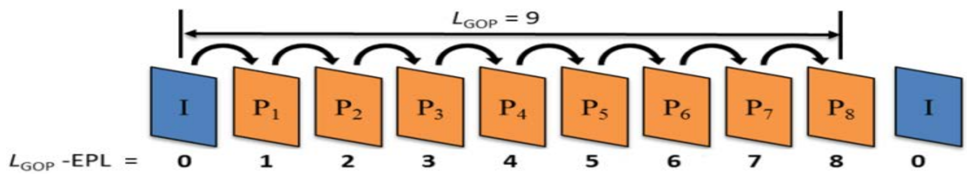

Figure 2. The EPL Video Frame Importance Scheme for a GOP of Period *nine* frames: {$IP_1P_2P_3P_4P_5P_6P_7P_8$}

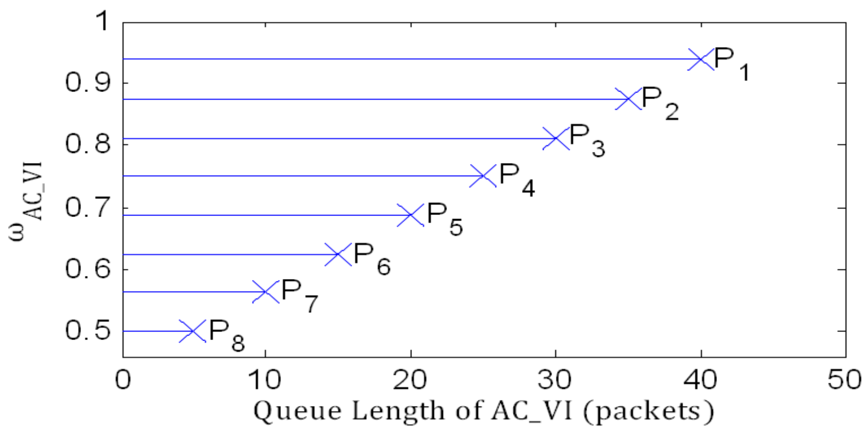

Figure 3. PWD: Prioritized Weighting Factors and Dropping Thresholds of AC_VI

transmission resources for AAC_VI. As a result, it seems that one can only gain some benefits for one AC queue by sacrificing those for the other.

Our proposed PWD design can provide a *win-win game* for both AC_VI and AAC_VI. Firstly, it is based on a cross-layer design which combines a *real-time* video packet importance scheme called *error propagation length* (EPL) [12] in the Application layer and a *priority-based weighting and dropping* algorithm for video packet transmissions in the MAC layer.

A typical example of the EPL scheme for a *group of pictures* (GOP) with a period of nine video frames is illustrated in Figure 2, where the importance level of each video frame can be represented by its EPL value, running within {9, 8, 7, 6, 5, 4, 3, 2, 1} respectively for {I, P1, P2, P3, P4, P5, P6, P7, P8}. Note that the packets associated with the same video frame share the same importance level.

The algorithm of PWD is explained in Figure 3, using Figure 2 as a typical example. Firstly, as aforementioned, the weighting factor of AC_VI (denoted as ω_{AC_VI}) should be larger than that of AAC_VI (denoted as ω_{AAC_VI}). In other words, ω_{AC_VI} should be within the range [0.5, 1] while ω_{AAC_VI} within the range [0, 0.5]. The definition of ω_{AC_VI} can be found in Equation (1), where L_{GOP} is the period of GOP, and α is a chosen constant so that the range [0.5, 1] is equally spaced by the prioritized values of ω_{AC_VI}: the larger the EPL, the larger the ω_{AC_VI}. In other words, $\omega_{AC\_VI} = 1$ for I packets, $\omega_{AC\_VI} = 0.5$ for P8 packets. On the other hand, the value of ω_{AAC_VI} is dependent on that of ω_{AC_VI} in the sense that their sum needs to meet the normalization condition, as shown in Equation (2).

In order to achieve graceful degradation of video transmission quality in bandwidth shortage, another concept called *priority dropping* also comes in to play for both AC_VI and AAC_VI, where a set of *dropping threshold* (DT) values are introduced. As a result, when the queue length of AC_VI increases, P8 packets will be discarded at the earliest time while P1 packets at the latest moment, requiring that no I packets be dropped at all until full queue occurs. Similarly for AAC_VI. Equation (3) defines the values of DT, where β is a constant which keeps DT run within the range between zero queue length and the *buffer limit* (BL) of AC_VI or AAC_VI, as indicated by the 'X' marks in Figure 3, where $\beta = 5$ for $L_{GOP} = 9$. Note that S is a switch value: 0 for the I packets, and 1 for all the P packets.

$$\omega_{\text{AC\_VI}} = 1 - \alpha(L_{GOP} - EPL) \quad (1)$$

$$\omega_{\text{AAC\_VI}} = 1 - \omega_{\text{AC\_VI}} \quad (2)$$

$$DT = BL - \beta\{(L_{GOP} - EPL) + S\} \quad (3)$$

3. Experimental Results

3.1. Simulation Setup

The experiments were conducted for H.264 video streaming over an 802.11aa wireless ad-hoc network, based on our implementation for the ICAP and SCS scenarios of 802.11aa under the *ns*-2 network simulation environment [13]. Figure 4 shows the adopted simulation topology with network capacity of 1 Mbps, and thus the end-to-end bandwidth is expected to be smaller than 1 Mbps due to the *fixed* and *random* backoff mechanisms of MAC layer. Two identical video streams were sent *asynchronously* from the *video* sender to *video* receiver, one of which was sent via AC_VI *with* the *real-time delay* constraint while the other of which via AAC_VI *without* the constraint. Concurrent cross-traffic flows, detailed below, were sent from both the two senders (the *video & cross-traffic* sender and the *cross-traffic* sender) to their corresponding receivers. Note that both the *video* and *cross-traffic* types are unicast-based.

Two reference video sequences (*NEWS QCIF* and *Carphone CIF*, both in the YUV format) [14] were adopted as the video sources. Each video sequence was coded into a 384-kbps bit-stream in the GOP pattern $\{IP_1P_2P_3P_4P_5P_6P_7P_8\}$ with a period of nine video frames by the H.264 reference software (JM 10.2) [15]. The video transmission quality evaluation toolset *EvalVid* [16] was used to packetize the bit-stream and the video packets were then sent over the above simulation topology. As aforementioned, two bit-streams coded from the same video sequence were sent via AC_VI and AAC_VI respectively for *real-time* and *non-real-time* streaming flows.

The performances of both the *real-time* and *non-real-time* video streams were evaluated under *six* congestion cases, denoted as *n* running from 1 to 6 to represent increasing congestion levels caused by concurrent *cross-traffic* flows. For $n = 1$, the *cross traffic* was formed by one VoIP flow of 16 kbps via AC_VO, as well as one UDP flow of 10 kbps via AC_BE and one TCP flow via AC_BK (recall that TCP is capable of *flow control*). Accordingly, the number of *cross-traffic* flows was increased to be of 6 times when $n = 6$.

Tphe quality of each video frame was evaluated based on the *yield* part of *peak-signal-to-noise-ratio* (Y-PSNR), as defined by Equation (4), where $(Y_{i,j} - Y^*_{i,j})$ is the *yield* difference per pixel between *before* and *after* quality impairment. The performance metric for the entire video sequence was adopted to be an average over all the frame-based Y-PSNR values. To be statistically meaningful, note that the performance data point for each congestion case was averaged over 30 simulation runs, denoted as Average Y-PSNR.

$$\text{Y-PSNR (dB)} = 10\log_{10}\frac{255^2 MN}{\sum_{i=1}^{M}\sum_{j=1}^{N}\left(Y_{i,j} - Y^*_{i,j}\right)^2} \quad (4)$$

Figure 4. Simulation Topology

Table 1 compares the adopted weighting factors and packet dropping conditions of our PWD design and the others. Note that the DEI bit of SCS-WRR starts to drop P packets when AC_VI has more than 20 packets or AAC_VI has more than 40 packets, considering the *real-time delay* constraint on AC_VI. Note that the same *buffer limit* of 50 packets for both AC_VI and AAC_VI was adopted.

Table 1. AC-specific Resource Parameters

Design	Weighting factors		Packet dropping	
	ω_{AC_VI}	ω_{AAC_VI}	AC_VI	AAC_VI
IACP-RR	0.5	0.5	not applied	not applied
IACP-WRR	0.9	0.1	not applied	not applied
SCS-WRR	0.9	0.1	DEI	DEI
PWD	Priority	1- ω_{AC_VI}	Priority	Priority

3.2. Non-real-time Performance Evaluation for AAC_VI

The AAC_VI performances of different designs are evaluated and compared under various congestion levels, based on which design has *the least packet losses* so as to achieve the *highest* Average PSNR. As shown in Figures 5-a and 5-b, our PWD design takes the lead for all the congestion levels, where the gains of PWD over IACP-RR (the second place) are in general slightly larger in *Carphone CIF* than in *News QCIF*. In addition, the fact that IACP-RR outperforms both SCS-WRR and IACP-WRR is well expected because WRR gives a much less probability for AAC_VI to be scheduled for the EDCAF function in front of the *double-queues* structure. Note that SCS-WRR is slightly better than IACP-WRR since the concept of SCS introduces the DEI bit for *graceful degradation*, as aforementioned.

3.3. Real-time performance evaluation for AC_VI

Unlike AAC_VI, the AC_VI performances of different designs should be evaluated and compared based on which design can not only receive the most video packets, but also achieve the highest *packet survival rate* after a *real-time cut.* Figures 6 and 7 show the performances of different designs under various congestion levels *before* and *after* the adopted *real-time cut* at 300 ms respectively. Note that IACP-RR is always the loser in both the cases since it is not suitable at all for *real-time streaming* via AC_VI. *Before* applying the *real-time cut*, there is no much difference (less than 1 dB) between our PWD design and SCS-WRR or IACP-WRR. *After* the real-time cut, however, the difference is amplified particularly in *heavy* congestion levels where our PWD design takes the lead. This shows the true power of our PWD design, *i.e.*, a *win-win game* for both AC_VI and AAC_VI in the congestion control of heavy traffic.

3.4. Sensitivity of real-time Cuts

The sensitivity of real-time cuts can further be examined over the range [200 ms, 400 ms], where any delay beyond 400 ms will be human-eye-perceptible. As shown in Figures 8-a and 8-b, taking n = 5 as a typical example, it is clearly seen that our proposed PWD design takes the lead over a large region of the examined range, where the *real-time* capability of *News QCIF* seems to be much better than that of *Carphone CIF*.

3.5. Effect of packet delay distribution

A deeper insight of the above *real-time* performance gains in heavy congestions can be obtained through the comparison of *packet delay distributions* in different designs, as shown from Figures 9 to 12, taking the n = 5 case of *News QCIF* as a typical example. Note that the vertical axes of these figures are in the *logarithmic* scale, it is thus the *height* that can closely represent the number of packets, not the *area size*.

The *packet delay distributions* of *received* video packets can be divided by the *real-time cut* at 300 ms into two parts: (1) *received-and-survived*, and (2) *received-but-cut-away*. The following three ratios, all normalized to the *total number* of video packets in the adopted reference video sequence, can thus be defined as another performance metric at the packet level.

- r_R: ratio of *received* packets before the real-time cut
- r_{RC}: ratio of *received-but-cut-away* packets by the real-time cut
- r_{RS}: ratio of *received-and-survived* packets after the real-time cut

It is clearly seen that our PWD design can not only achieve the highest value in r_{RS} (65%) but also the lowest value in r_{RC} (13%), which supports its leading place in Average Y-PSNR. It is interesting that although both IACP-WRR and SCS-WRR have higher values in r_R than PWD, roughly 3/5 and 2/5 of their *received* packets have been cut-away respectively. This also shows the true power of PWD in terms of *real-time design*.

4. Conclusions

In this paper, we have proposed a novel *double-queues* scheduler design, called PWD, for the newly released QoS protocol from IEEE 802.11aa-2012. Based on cross-layer design, PWD takes the EPL video importance scheme to form *priority weighting* for the *primary* queue, and *priority dropping* for both the *primary* and *alternative* queues. The simulation results show that the proposed design outperforms the conventional ones for both the *real-time* and *non-real-time* video streams via the *primary* and *alternative* queues respectively. Further insights from the *packet delay distributions* of different designs and the *real-time cut sensitivity* also support the superiority of PWD. In other words, our PWD design can actually achieve a *win-win game* for both *real-time* and *non-real-time* video streams.

Acknowpledgements

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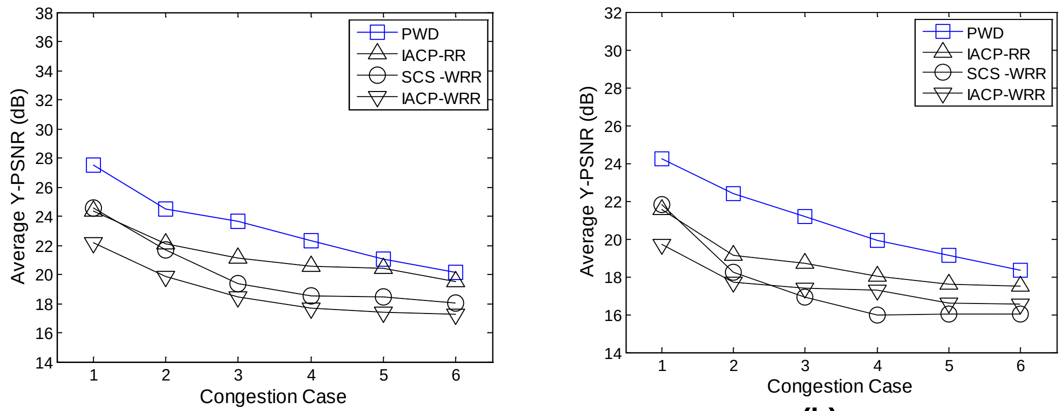
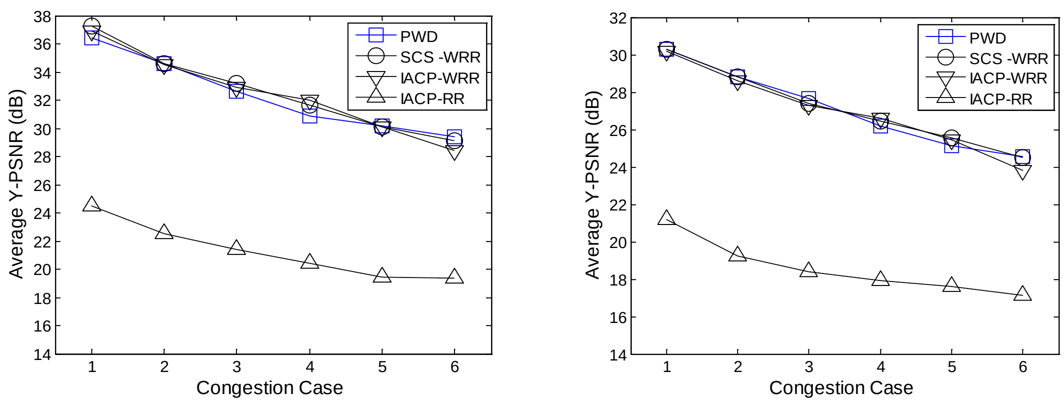


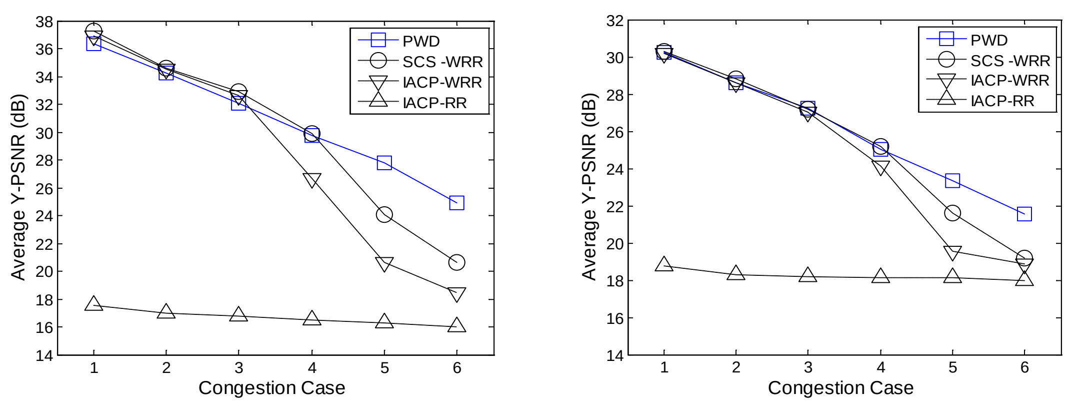
(a) (b)

Figure 5. Performance of AAC_VI versus Congestion Case: (a) *News QCIF*, and (b) *Carphone CIF* (*non-real-time*)



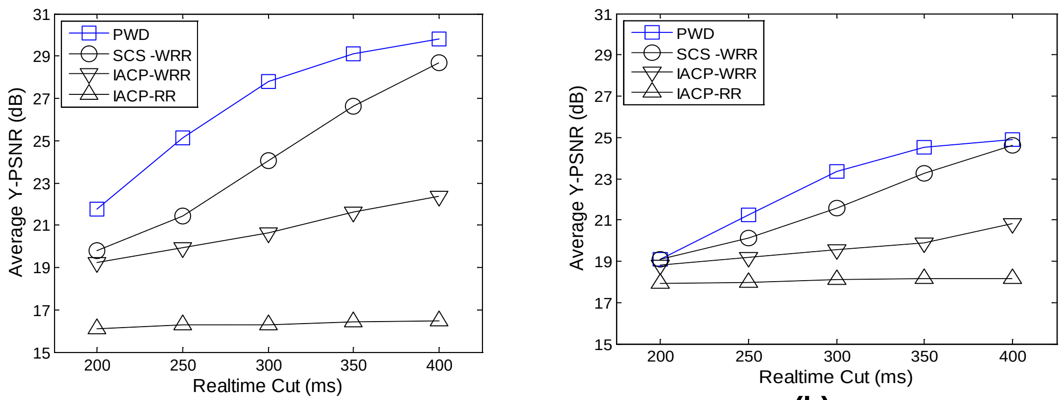
(a) (b)

Figure 6. Performance of AC_VI versus Congestion Case: (a) *News QCIF*, and (b) *Carphone CIF* (*before any real-time cut*)



(a) (b)

Figure 7. Performance of AC_VI versus Congestion Case: (a) *News QCIF*, and (b) *Carphone CIF* (*after the real-time cut at 300 ms*)



(a) (b)

Figure 8. Sensitivity of various *real-time* cuts: (a) *News QCIF*, and (b) *Carphone QCIF*

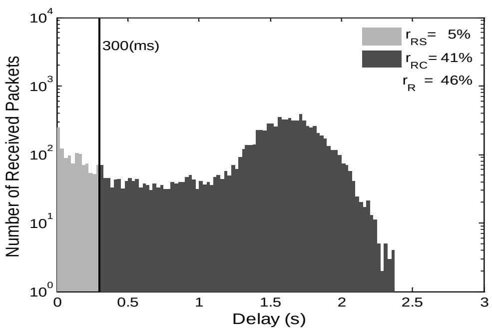


Figure 9. Packet Delay Distribution of IACP-RR

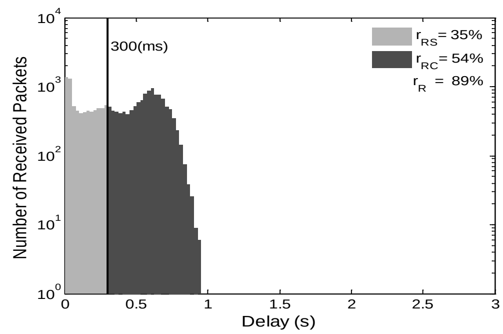


Figure 10. Packet Delay Distribution of IACP-WRR

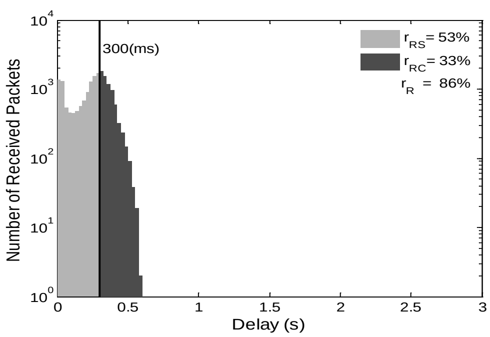


Figure 11. Packet Delay Distribution of SCS-WRR

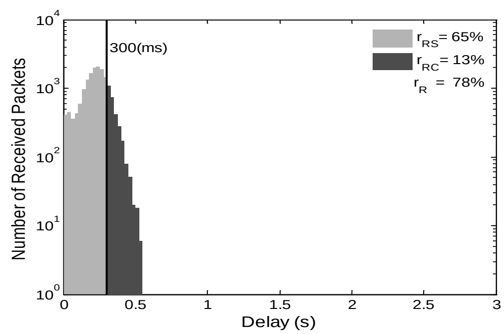


Figure 12. Packet Delay Distribution of PWD (the proposed design)

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