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User-Level Quality Assessment of a Delay-Aware Packet Dropping Scheme for VoIP

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Abstract

To deal with congestion control in internet protocol (IP) networks, different active queue management schemes have been successfully proposed. However, many of these schemes do not take into consideration the particular Voice over Internet Protocol (VoIP) requirements, thereby potentially penalizing this traffic. In this paper, we propose and evaluate a dropping packet selection algorithm for queue management schemes. This algorithm, called *Drop-SwD*, protects VoIP traffic without affecting other User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) traffic. Additionally, to preserve the quality of the VoIP service, the proposed algorithm selects the lowest quality packets (with less impact on the subjective quality) to be dropped.

The application of the algorithm in different active queue management schemes is evaluated using objective and subjective quality metrics and intelligibility metrics as well. As a result, we show that *Drop-SwD* improves the final quality of the voice packets' flow, without degrading the performance of other traffic class, with respect to the packet dropping schemes of reference.

Keywords: AQM, Dropping, QoE, QoS, Service Differentiation, VoIP.

1. Introduction

The internet protocol (IP, *best-effort*) has expanded very rapidly becoming the convergence technology for interconnecting different networks. A great diversity of traffic types have emerged for which IP was not originally designed, such as multimedia traffic with real-time requirements. The internet is not always adequate to satisfy all of the requirements of these new network applications, and it often becomes necessary to incorporate mechanisms that ensure an adequate quality of service (QoS). Techniques that can differentiate between different applications and that can be stable and consistent are required to satisfy the demands of QoS, as well as those of Quality of Experience (QoE).

To implement differentiated services, the Internet Engineering Task Force (IETF) proposed the architecture of Integrated Services (*IntServ*, [1]) and Differentiated Services (*DiffServ*, [2]). The implementation of these models requires the massive deployment of *IntServ* and *DiffServ* routers, respectively. In addition, schemes such as active queue management (AQM, [3]) and scheduling algorithms have been also proposed.

For interactive voice services the time for recovering lost packets is limited. As the number of consecutive lost (or damaged) packets increases, the user will perceive a greater degradation in the quality.

Packet losses are usually caused by intermediate routers under severe congestion conditions. Even farther, under less severe network conditions, any VoIP packet that does not arrive to the destination user in the proper time period –defined by the play-out buffer- is considered unusable; in other words, it is also perceived as a loss. These packet losses are the consequence of the accumulated delay (propagation plus transmission) plus queuing delay caused by the slightly congested routers.

To mitigate congestion episodes, AQM schemes play a relevant role by marking or dropping packets at intermediate routers by using an increasing probabilistic function based on the average buffer occupancy.

In this paper, we propose a victim packet selection algorithm called *Drop-Sel with Delay* (*Drop-SwD*) for AQM schemes. Distinct to its predecessor [4], *Drop-SwD* considers the particular delay requirements of VoIP packets. *Drop-SwD* is designed to improve the use of network resources, incrementing the VoIP user perceived quality. For this goal, the estimated end-to-end packet delay is taken into consideration for victim selection in AQM.

Drop-SwD defines the packet drop priority according to the dynamic queue occupation by considering three traffic types. This mechanism reduces the probability of dropping valid (still usable) packets of the VoIP flows in the presence of invalid packets (those with inacceptable accumulated delay). By including the end-to-end delay as a victim selection criterion, this mechanism aims to provide a differentiated service to the VoIP traffic without penalizing other traffics.

The benefit of *Drop-SwD* has been evaluated through simulations in the Network Simulator (ns-2). For comparison purposes, two reference victim selection schemes (*Drop-Tail and Drop-Front*) are also simulated. The results indicate that *Drop-SwD* improves the VoIP traffic performance in terms of reducing usable packet dropping probability, MOS scores and also improves the end-user intelligibility.

The remainder of the document is organized as follows. In section 2, the state of the art is analyzed by fundamentally evaluating the AQM schemes that provide a differentiated service. Section 3 details the characteristics of the *Drop-SwD* victim selection algorithm. Section 4 discusses how to estimate the usefulness of the queued VoIP packets by assessing the experienced delay. The simulation scenarios and configuration details are specified in section 5. In section 6, the impact of *Drop-SwD* on VoIP, in terms of the loss rate and packet delay, is analyzed. Section 7 offers an analysis of the selection scheme performance in terms of the perceived quality offered by the final user using E-Model and an automatic speech recognizer. Finally, the document is summarized in section 8.

2. Related Works

One of the main challenges that IP technologies face is the prevention of network congestion. To deal with this problem, a number of prominent AQM systems have been proposed in the last few years. Among them the *Random Early Detection* (RED) [5] scheme is considered to be a reference seminal work. Some AQM schemes have been created not only with the intent to detect, avoid and control congestion but also with the objective to improve the QoS (packet loss, link utilization, delay and jitter). For this latter objective, schemes such as Extension of Hybrid RED (XHRED) [6], Weighted RED (WRED) [7] and Weighted RED with Threshold (WRT) [8] employ different levels of dropping probability according to the previously established priority of the packet. Rainbow Fair Queuing (RFQ) [9] provides multiple levels of dropping, without considering per-flow-state information. However, if reactive and non-reactive (TCP and UDP traffic) flows are mixed in a simple queue, these schemes prevent or avoid congestion independent of the reactive nature of the involved traffic. To solve this problem, some schemes have incorporated a dropping probability that is attached to the requirements or the class of the type of traffic that they belong to. Dynamic-Class Based Thresholds (D-CBT) [10] apply different policies when classifying flows into three classes, TCP, UDP (multimedia) and other UDP. In this case, D-CBT applies the same dropping probability to VoIP as to other multimedia flows. When congestion occurs, PUNSI [11] adopts two different dropping strategies depending on the traffic type (UDP or TCP). PUNSI is more aggressive with non-reactive flows (including VoIP), which allows for the allocation of more bandwidth to the reactive flows. Similar to PUNSI, Jitter Detection (JT) [12] classifies packets into two possible categories: TCP or UDP. The UDP traffic is subject to the JD scheme, while the TCP is subject to the RED standard. JT aims to reduce the average delay of the multimedia packets. RED-Boston [13] was developed to satisfy the QoS requirements of all applications by introducing a suggested packet delay. The suggested delay indicates the degree of delay that the application is able to tolerate without affecting its performance.

In contrast to these schemes, *Drop-SwD* considers both the specific application types and a service designed for VoIP flows that adapts to the network load dynamic, and particularly to the queuing delay fluctuations.

The LIBS scheme [14][15] was recently developed; to prioritize some traffics (for example VoIP) this scheme considers only the size of the packet when classifying the packets into different traffic classes. Similar to LIBS, *Drop-SwD* seeks to provide a QoS application-oriented strategy designed for producing a dynamic and differentiated service by considering the requirements of VoIP services.

3. The *Drop-Sel with Delay* Victim Selection Algorithm

The dropped packet selection algorithms used by most of the AQM schemes select the last, first or random packet in the queue. Recently, in [4], another variant of victim selection, called Selective Dropping (*Drop-Sel*), was introduced. *Drop-Sel*, in contrast with traditional selection schemes, includes the traffic type to which the packet belongs and provides a fair service when reactive and non-reactive sources are mixed in a simple queue. Given that the interactivity demands of the real-time service are mainly linked to the loss rate and experienced delay, proper victim selection could improve the impact with respect to QoS and QoE if it considers the traffic type and the delay experienced by the VoIP packet.

In this research, an improved victim selection algorithm called *Drop-Sel with Delay (Drop-SwD)* is proposed. This algorithm is motivated by the limitation in the end-to-end delay that interactive VoIP traffic demands. If a packet that belongs to an interactive audio flow experiences an end-to-end delay (typically, greater than 300 ms), this packet becomes unusable (invalid packet), thereby wasting network resources. Therefore, a packet is defined as valid if its age, defined as the time elapsed since its origin, never exceeds the maximum alive time permitted. Under heavy traffic or congestion conditions, a packet may become invalid on its path and thus unnecessarily use memory and bandwidth until getting its destination. Even worse, given the involvement of a router, if a victim selection criterion is not taken into consideration, the AQM algorithm can drop a valid voice packet while keeping an invalid packet in the queue.

Similar to the study reported in [4], in this paper, we rely on the determination of the type of traffic that most greatly contributes to the congestion problem. To this end, we take into consideration three kinds of traffic: real-time non-reactive UDP (VoIP flows), other non-real-time UDP flows and finally TCP. Contrary to the previous work, *Drop-Sel with Delay* considers a criterion based on the utility of the packet when the type selected as a victim corresponds to the real-time type (VoIP).

Specifically, when the AQM decides to drop a packet, the reference functions of the *Drop-Sel* are performed and determine which of the three traffic types occupy the most space in the queue. If the other UDP or TCP type is using the most memory space during the congestion periods, the *Drop-SwD* algorithm looks for a packet corresponding to the selected type that is the closest at output queue and drops it. However, if the real-time type occupies the most space in the queue, the algorithm switches to another traditional strategy for victim selection (in this case, *Drop-Tail*). Then, the algorithm performs the functions included in *Drop-Tail*, i.e., the direct selection of the packet that has just arrived. If the packet that has just arrived belongs to the other UDP or TCP type, it is dropped. In contrast to *Drop-Sel* and *Drop-Tail*, if the packet about to leave the queue belongs to the real-time traffic type, the algorithm evaluates all the packets in the queue and identifies the audio packet that is considered invalid and that is the closest at output queue. In the absence of invalid audio packet (>300 ms), it simply drops the audio packet closest at output queue. Therefore, it behaves as reference *Drop-Sel*. In this investigation, we assume that the audio frames are encapsulated in a RTP/UDP packet.

Every time a packet is queued or removed from the queue, the *Drop-SwD* algorithm conveniently updates the counters. In Fig. 1, the *Drop-SwD* procedure is summarized in a simple flow-chart.

As we have observed, to avoid wasting network resources, *Drop-SwD* identifies and drops invalid packets in the queue that belong to the VoIP traffic type. These invalid packets will be penalized to protect the valid packets. As soon as the non-usable audio packets are detected and dropped, the loss of valid packets can be reduced. Therefore, better QoS and QoE can be expected for the VoIP traffic.

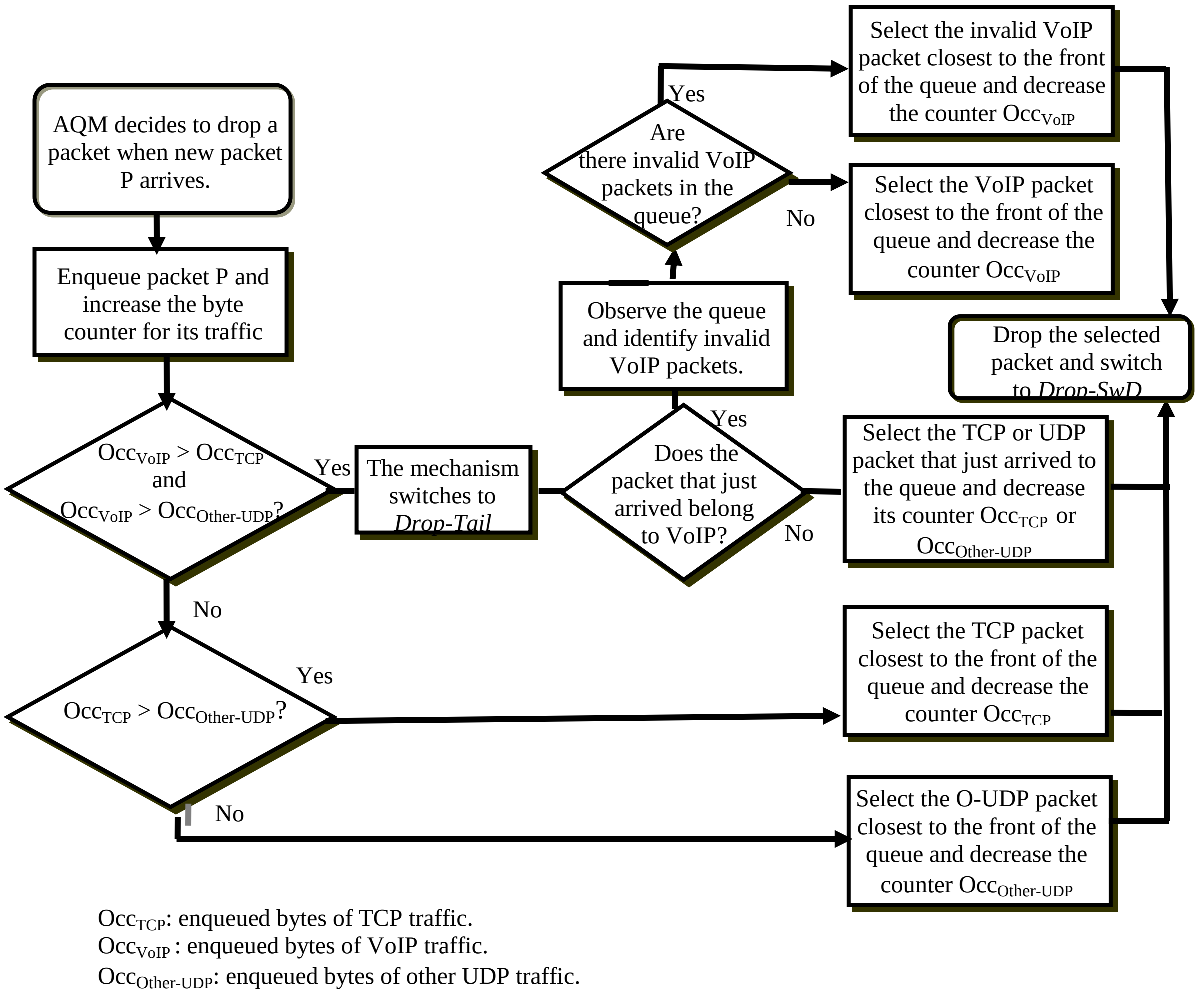


Figure 1. *Drop-Sel with Delay* flow diagram.

4. Selection Strategy Oriented Toward VoIP

A criterion to select the VoIP packet based on the experienced end-to-end delay has been incorporated into *Drop-SwD*. To approximate this value inherent to the algorithm, we propose schemes that determine the age or cumulative delay of the packet using a message from the RTCP protocol [16]. The source node, AQM router and destination node are assumed to be synchronized.

As can be observed in Fig. 2, the end-to-end delay of the packet in one direction can be modeled as the sum of two variables, the delay from the source node to the AQM router (denominated d_1) and the delay from the AQM router to the destination node (denominated d_2). The packet reaches the AQM router and the destination node with a transmission and queuing delay caused by manipulation through the links. The main idea behind *Drop-SwD* is to estimate the cumulative delay caused by each of the VoIP packets when these packets reach their destination node. This estimation enables the prediction of the possible packets that will be invalid.

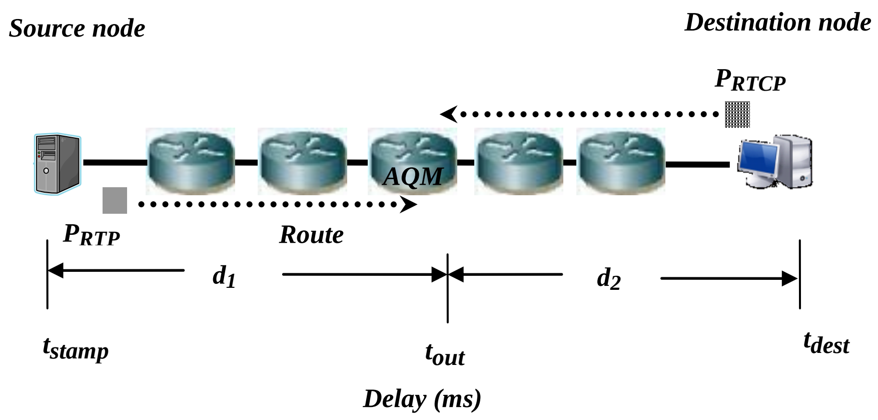


Figure 2. End-to-End delay scheme in one direction.

For this algorithm to be effective, in contrast to the Drop-Sel and Drop-Tail reference schemes, it will capture specific information to calculate the cumulative delay through the length of a network route.

4.1 Source Node to the AQM Router Delay d_1

To efficiently calculate the delay d_1, we suggest dividing this variable into two components. First, we propose the estimation of the packet delay from the source node to its position in the AQM queue (denominated d_{1a}). Second, the delay that will be incurred by the remaining time to be spent in the queue (denominated d_{1b}) should be calculated. *Drop-SwD* uses information from the packet and from the router itself. Specifically, the timestamp of the queued packet (denominated t_{stamp}) and the local time spent in the router (denominated t_{localq}) is used in the estimation of the delay d_{1a} for the queued packet in position i, as shown in the following expression:

$$d_{1a} = t_{localq} - t_{stamp} \tag{1}$$

For each packet that leaves the queue, the scheme obtains the generated queuing delay using the following expression:

$$d_{queue} = t_{outq} - t_{inq} \tag{2}$$

where t_{inq} is the arrival time of the packet to the AQM queue and t_{outq} is the output time from the queue. In Fig. 3, the different variables used by the algorithm are detailed.

By monitoring the queuing delay, we obtain the necessary information to characterize the network dynamic. With this value (d_{queue}, (2)), the *Drop-SwD* victim selection algorithm will calculate the time that a packet will wait in the queue before leaving. To perform this estimation, we will consider two additional variables: the amount of packets in the queue (denominated s_q) and the position of the packet in the queue (denominated i). The delay d_{1b} approximation can be expressed as follows:

$$d_{1b} = \left(\frac{d_{queue}}{s_q}\right) * (i-1) \tag{3}$$

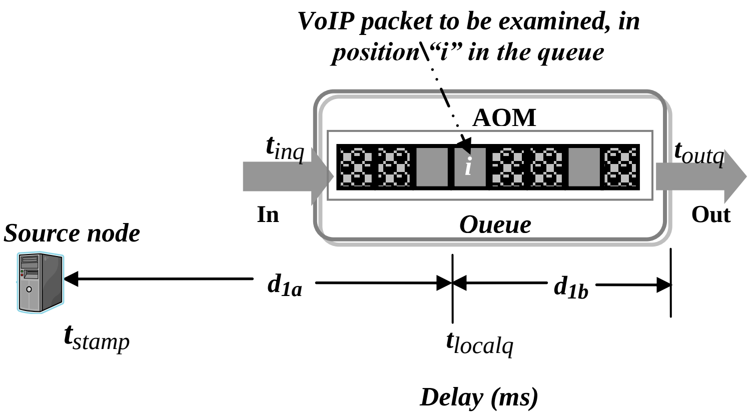


Figure 3. Queuing delay scheme.

4.2 Delay from the AQM Router to the Destination Node, d_2

For each VoIP packet that leaves the AQM queue, *Drop-SwD* inserts the output time of the packet into the RTP header. This value will allow for the estimation of the last of the end-to-end delay variables, specifically d_2 (see Fig. 2). Once the destination node receives an RTP packet, it examines the header fields of the packet and calculates the cumulative delay between the AQM router and itself, thus determining the difference between the two specific times:

$$d_2 = t_{dest} - t_{outq} \tag{4}$$

where t_{dest} indicates the time that the packet was received at the destination node and t_{outq} indicates the time that the packet left the AQM queue. The obtained measurement (d_2, (4)) is fed back to the source node through the "receiver report" RTCP messages.

When the source node receives the RTCP control message, the measured delay is extracted and inserted into the following RTP packets to be transmitted. As soon as the source node receives a new RTCP report, the new delay measurement d_2 is obtained. With frequent control messages arriving at the source node, up-to-date information will be obtained from the packet, and an appropriate delay will be determined for the load fluctuations in the network.

Under congestion conditions, if the packet accumulates an end-to-end delay of greater than 300 ms ($d_{1a} + d_{1b} + d_2 > 300$, expressions (1, 3 and 4)), it will be considered invalid.

In Fig. 4, we provide an example of the *Drop-SwD* victim selection procedure. We assume that the VoIP traffic uses the highest router memory and that it has a packet arriving at the time at which the AQM will drop a packet. The victim selection scheme evaluates the delay experienced by each of the queued audio packets. The queued packet was found to have a delay greater than the established threshold (300 ms).

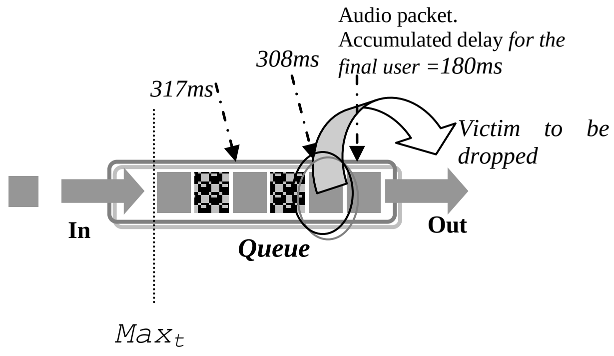


Figure 4. Example of the *Drop-SwD* victim selection.

For example, suppose that a packet accumulates 317 ms and another 308 ms. In this case, *Drop-SwD* will drop the packet with 308 ms (the closest one at output queue) instead of the packet that has accumulated the greatest delay (317 ms). If the selected victim were the other packet, the 308 ms packet may eventually be sent to the destination node (before being able to drop it), which could unnecessarily waste resources. *Drop-SwD* attempts to avoid transmitting packages that cannot be used by the user.

5. Simulation

To measure the impact of *Drop-SwD* on the service offered to the VoIP flows, specifically over the QoS and QoE, a number of simulations were executed using *Network Simulator* (*ns-2*) [17]. Two selected AQM schemes (RED [5] and REM [18]) were configured using the victim selection algorithm and evaluated under several levels of congestion. For comparison reasons, the *Drop-Tail* and *Drop-Front* were also applied to the AQM schemes.

5.1 Scenario

For the service configuration, we considered the standard single-bottleneck *dumbbell* topology shown in Fig. 5. In this network scenario, a number of TCP, VoIP (UDP) and other UDP non-reactive flows compete for resources in the router AQM (R0).

In the scenario, R0 is assumed to have an AQM scheme with *Drop-Tail*. The router R0 was stressed with four different workloads. The set of workloads percentages of each type of traffic are summarized in Table 1 (labeled as CA1, CA2, CA3 and CA4). Our intention is to verify the impact of the selection schemes under different levels of multimedia traffic.

Assuming that the different flows are not synchronized, a number of congestion periods will be randomly generated.

Table 1. Workloads of UDP and TCP Traffic Flows.

Case	VoIP	Other UDP	Total UDP	TCP
CA1	47%	36%	83%	17%
CA2	54%	19%	73%	27%
CA3	13%	52%	65%	35%
CA4	29%	6%	35%	65%

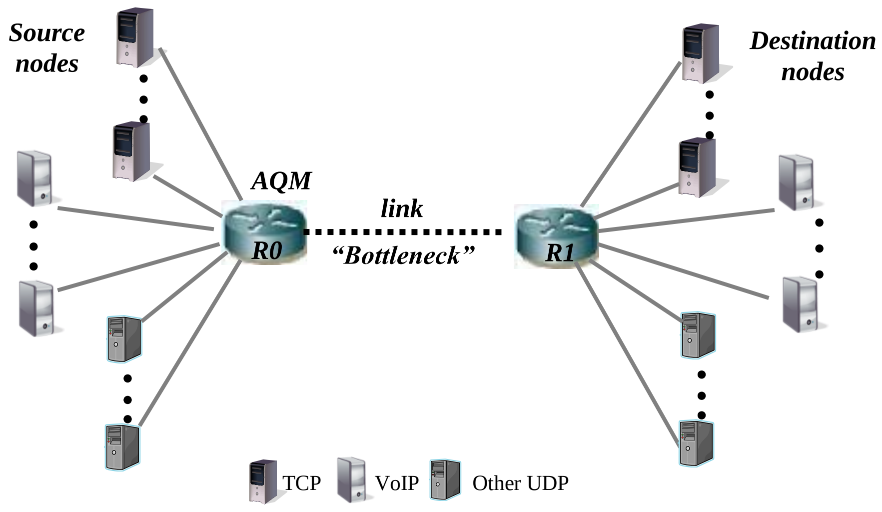


Figure 5. *Dumbbell* topology.

5.2 Traffic Configurations

A number of FTP sources generate TCP segments configured with a length equal to 1500 bytes, and the VoIP sources generate RTP packets (encapsulated inside UDP datagrams) that stand for G.711 encoded voice [19]. The size of the non-VoIP UDP packets is also 1500 bytes. All of the sources were modeled using ON/OFF traffic pattern. The ON period for the VoIP application lasts 180 seconds, while the OFF period lasts 100 seconds [20]. The FTP traffic follows a Pareto distribution with a shape parameter of k=1.4, an average ON period equal to 2 seconds and an OFF period that follows an exponential distribution with an average duration of 1 second [21]. Finally, for the other UDP applications, the ON period lasts 300 seconds, and the OFF period lasts 200 seconds. All the traffic sources start to send packages with a uniformly distributed probability at the first 15 seconds of the simulation.

In this investigation, the voice streams are generated using four VoIP applications (A, B, C and D) with different interpacket periods and different packet sizes. The VoIP flows that compete with the TCP and other non-reactive UDP sources in each case (see Table 1) are specified in Table 2. To measure the service impact proportional to the QoE, we evaluated victim selection strategies considering the cases for which links have large propagation delays. The established delays for each of the VoIP applications are shown in Table 2.

Table 2. VoIP Flow Specifications.

Application	Interpacket period	Packet size	End-to-end delay	Number of flows in each case			
				CA1	CA2	CA3	CA4
A	10 ms	92 *bytes*	39 ms	10 *(A1 to A10)*	10 *(A1 to A10)*	5 *(A1 to A5)*	2 *(A1 to A2)*
B	30 ms	252 *bytes*	79 ms	10 *(B1 to B10)*	10 *(B1 to B10)*	5 *(B1 to B5)*	3 *(B1 to B3)*
C	60 ms	492 *bytes*	169 ms	10 *(C1 to C10)*	10 *(C1to C10)*	5 *(C1 to C5)*	2 *(C1 to C2)*
D	30 ms	252 *bytes*	239 ms	10 *(D1 to D10)*	10 *(D1to D10)*	5 *(D1 to D5)*	3 *(D1 to D3)*

6. Results

6.1 VoIP Traffic Evaluation

To analyze the impact of the *Drop-SwD* scheme on the VoIP flows, we rely on the interactivity restrictions of the real time service. More precisely, we have examined the packet loss and experienced delay factors. First, the relation with packet loss was evaluated: drops in the AQM router (to report and prevent congestion) and drops at the destination node (due to the arrival of invalid packets). The results extracted for the different cases including the *Drop-SwD* procedure in AQM RED scheme are shown in Fig. 6-9. The results reflect the fact that if a victim selection criterion is taken into consideration, the resources of the network can be more efficiently used.

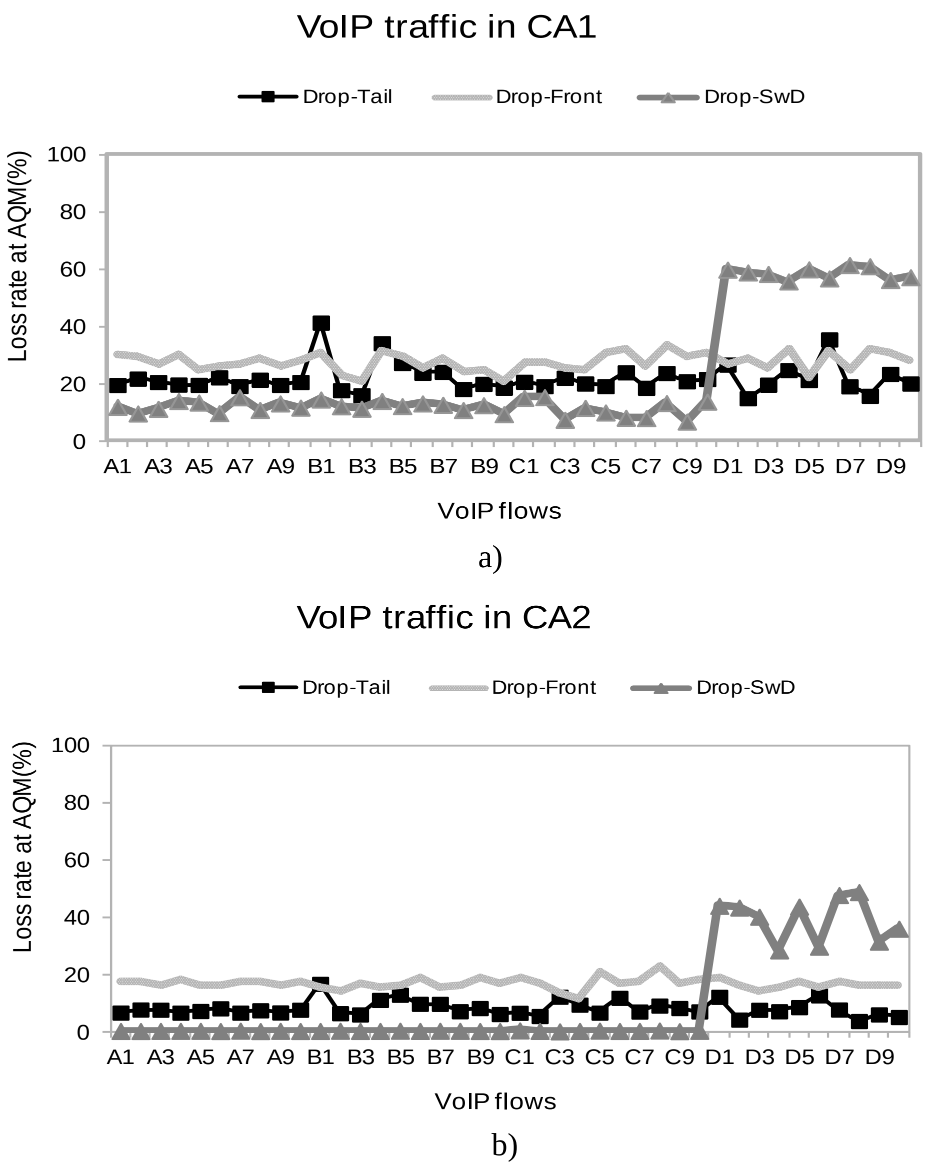
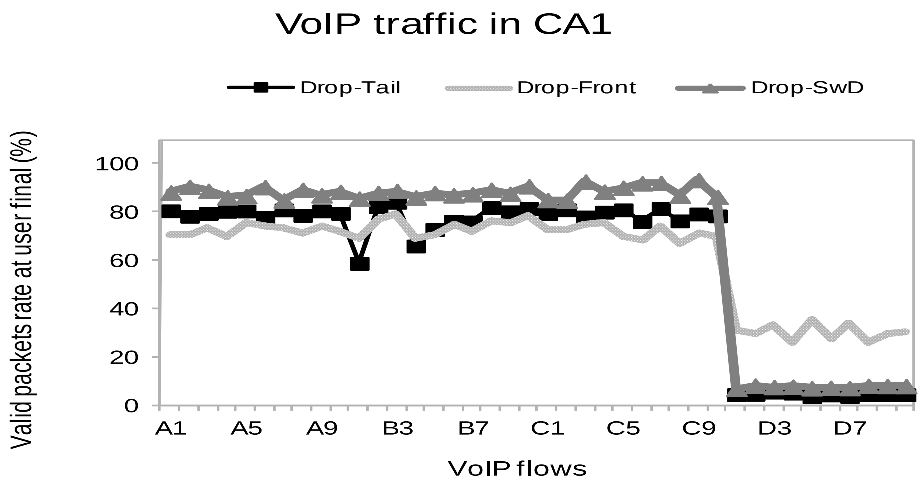


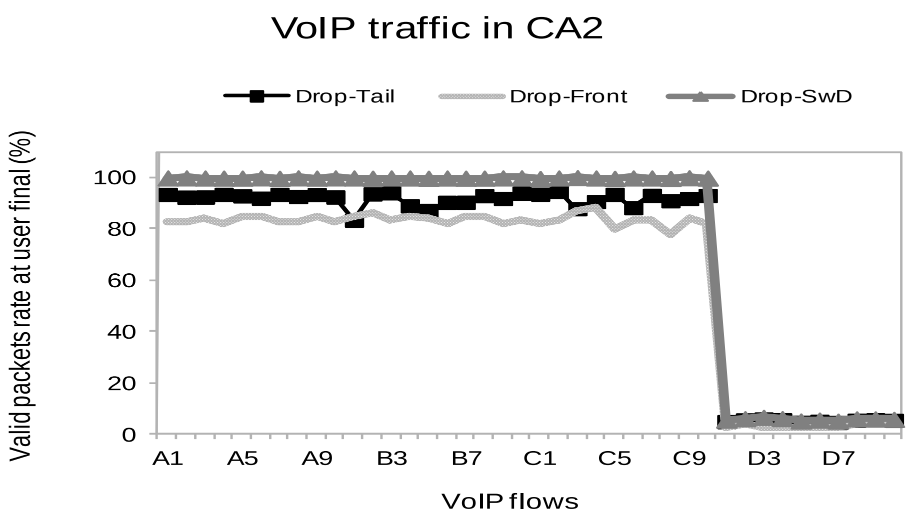
Figure 6. VoIP packets loss rate at AQM RED for cases CA1 and CA2.

With the main goal of avoiding resource waste, the *Drop-SwD* scheme has a priority to penalize (by dropping) the audio packets that are predicted not to arrive to the user within the expected time of 300 ms ($d_{1a} + d_{1b} + d_2 > 300$, *expressions (1, 3 and 4)*). In cases

CA1 and CA2, where TCP and other UDP flows compete with VoIP flows (dominant and responsible of the congestion), *Drop-SwD* detects that the flows designated D1 to D10 generate invalid packets. Therefore, the algorithm selects them as drop victims in the AQM (see Fig. 6a and Fig. 6b) instead of the packet generated by the application types A, B and C, which will not be rejected when they arrive at their final user. As soon as the unusable audio packets are detected and dropped in advance, the number of valid packets arrived at the destination node can be incremented, as is shown in Fig 7a and 7b. For example, if we compare the impact of *Drop-SwD* with that of *Drop-Tail*, the global rates of valid packets in the flows A, B and C (not affected by the delay) are increased from 78.23% to 88.32% in CA1 and from 91.74% to 99.81% in CA2. In contrast, the other selection schemes in RED which do not take into account the delay restriction allow higher arrival rates of invalid packets, typically of the flows that are most affected by the experienced end-to-end delay.



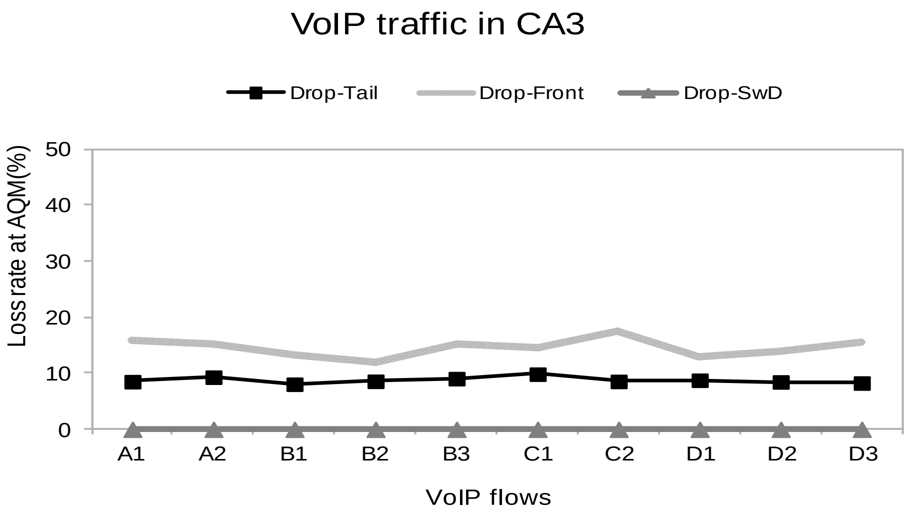
a)



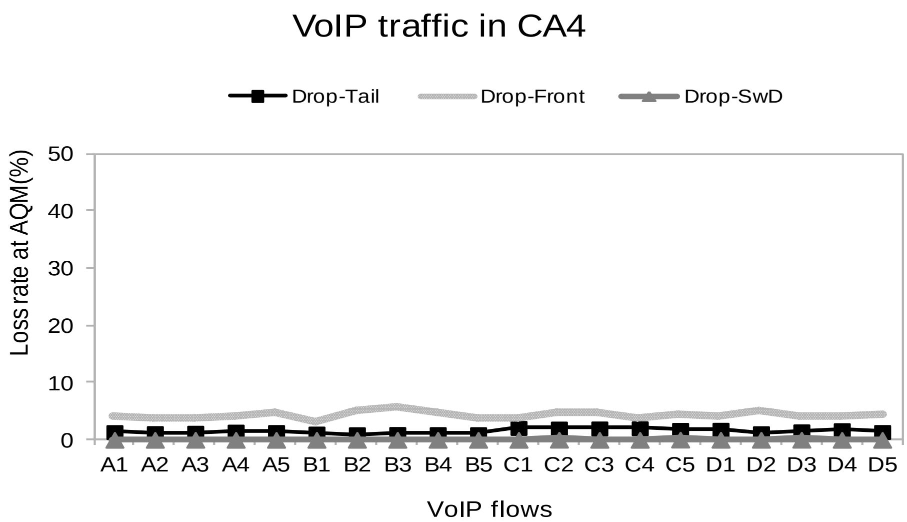
b)

Figure 7. Valid VoIP packets rate at user final for cases CA1 and CA2.

In cases CA3 and CA4 (see Fig. 8-9), the VoIP flows are not considered to be dominant. In these cases *Drop-SwD* detects that TCP and other UDP non-real-time flows are responsible for the congestion and consider them to be the first option for drop victims in the AQM. Therefore, the losses for the VoIP flows (see Fig. 8a and 8b) are significantly reduced. However, the VoIP selection strategy is not applied; note in Fig. 9a and 9b that the rate of valid packets increases even for the flows generated with application D, which is the most affected by the end-to-end delay. This behavior results from *Drop-SwD* selecting the correct traffic type as the drop victim and simultaneously reducing the average queuing delay of the router independent of the load conditions (due to the advantages of dropping the closest packets at output queue rather than those that has just arrived). The total losses are also reduced for all VoIP flows, thereby increasing the arrival rate of valid packets (see Fig. 9a and Fig. 9b). As an example, if we compare the impact of *Drop-SwD* with that of *Drop-Tail*, the valid packets global rates of the flow types A, B and C increase from 91.15% to 100.00% and those of flow type D increase from 34.43% to 73.93% in the case of CA3.

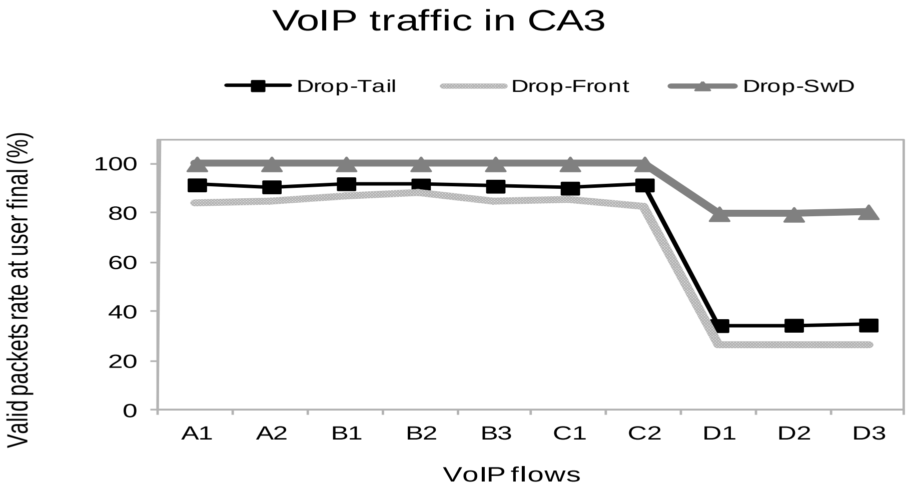


a)

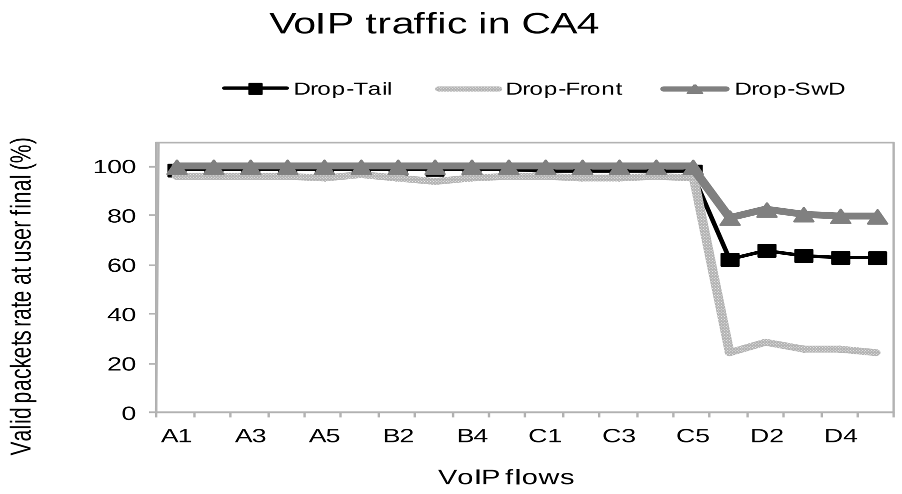


b)

Figure 8. VoIP packets loss rate at AQM RED for cases CA3 and CA4.



a)



b)

Figure 9. Valid VoIP packets rate at user final for cases CA3 and CA4.

Given that *Drop-SwD* uses a traffic classification and priority to establish packet selection strategies, we investigated its impact in another AQM scheme. During the experimental evaluation, we also included the selection algorithms in the REM scheme. The results, which are shown in Fig. 10-13, confirm that *Drop-SwD* provides a more efficient use of network resources compared with the other two selection algorithms.

When examining the behavior of the AQM REM with respect to RED, we observe that AQM REM provokes less queue congestion occurrences, thereby reducing the queuing time. However, the difference between the behaviors of the AQM schemes can be clearly observed in Fig. 10 and Fig. 12 because the implementation of *Drop-SwD* significantly reduces the losses in the AQM REM router for VoIP flows. Similar to RED, the reduction in the loss rate at the router and dropping the closest packets at output queue, increase the valid packet arrival rate at the destination node, as is shown in Fig. 11 and Fig. 13.

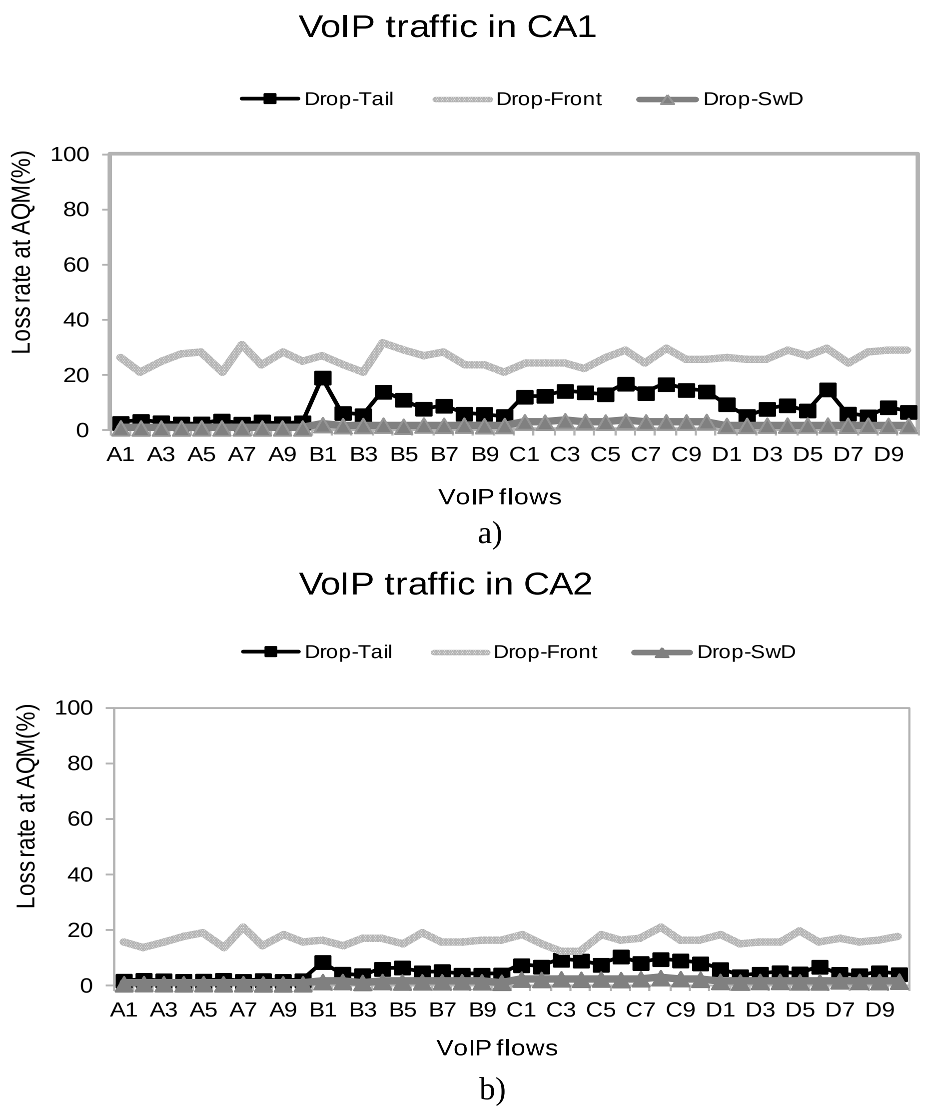


Figure 10. VoIP packets loss rate at AQM REM for cases CA1 and CA2.

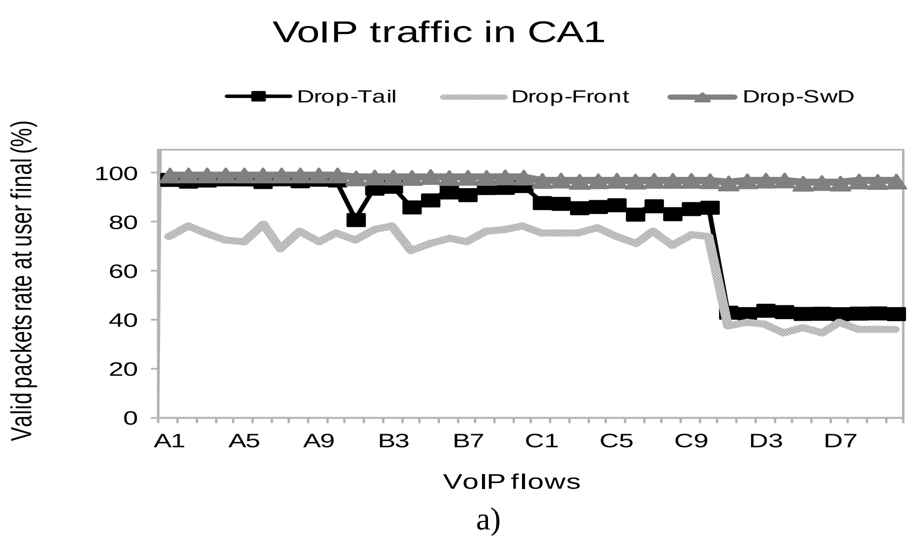
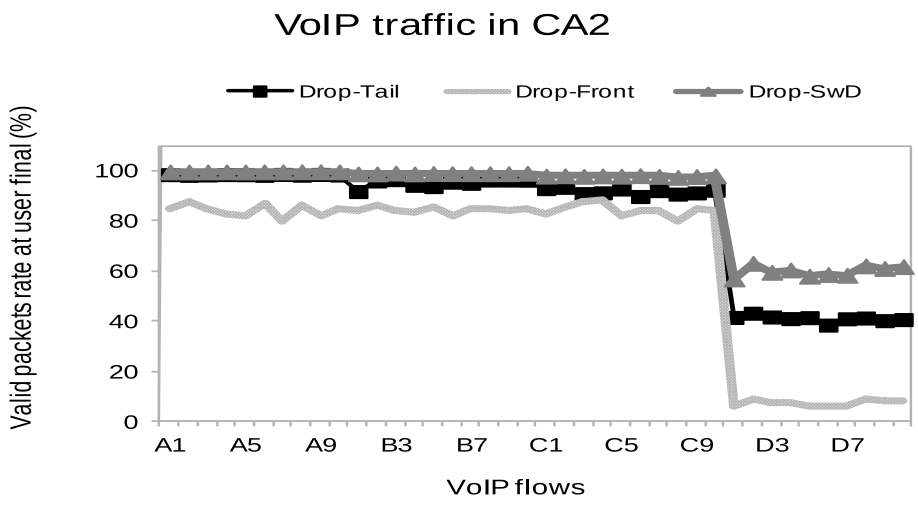
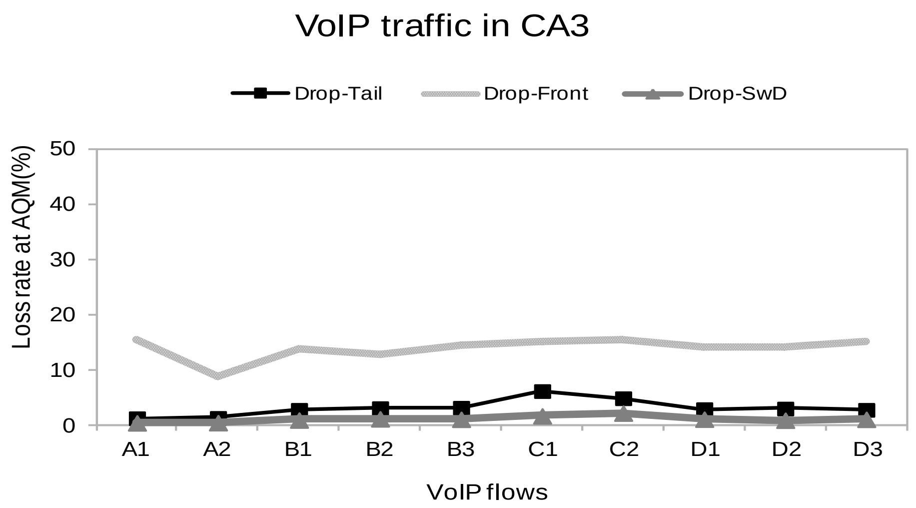


Figure 11. Valid VoIP packets rate at user final for case CA1.

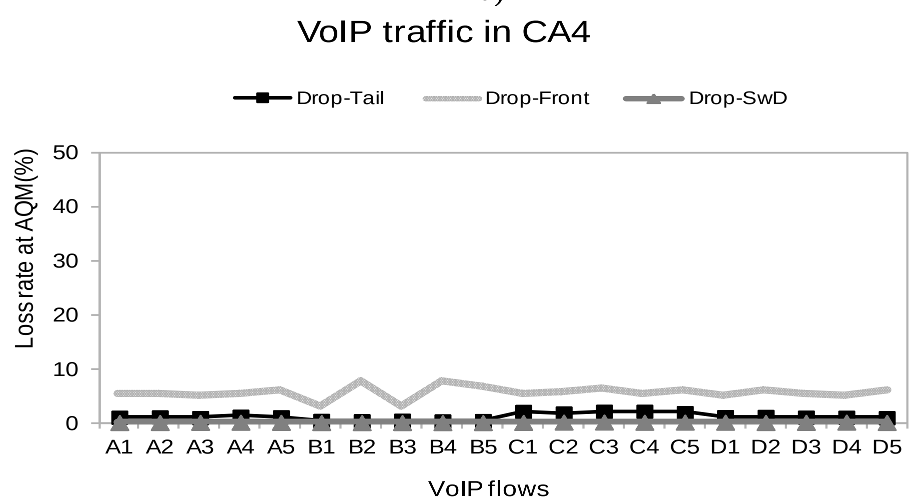


b)

Figure 11. Valid VoIP packets rate at user final for case CA2.

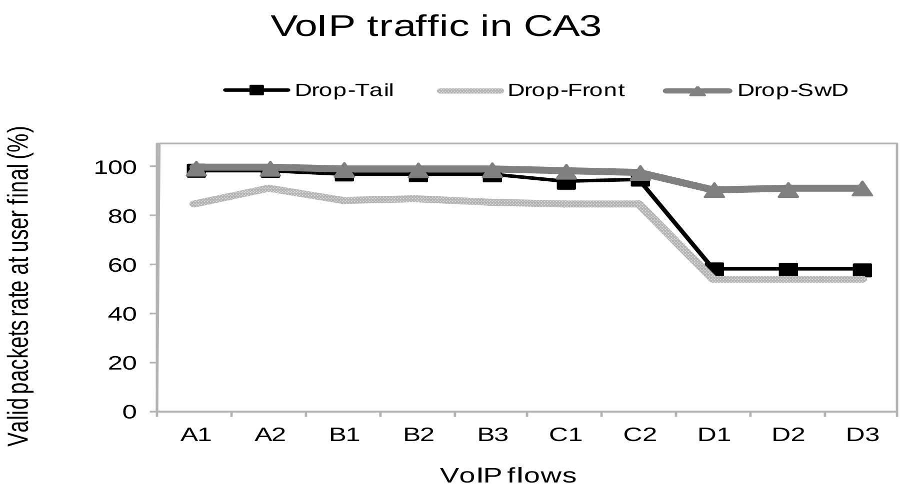


a)

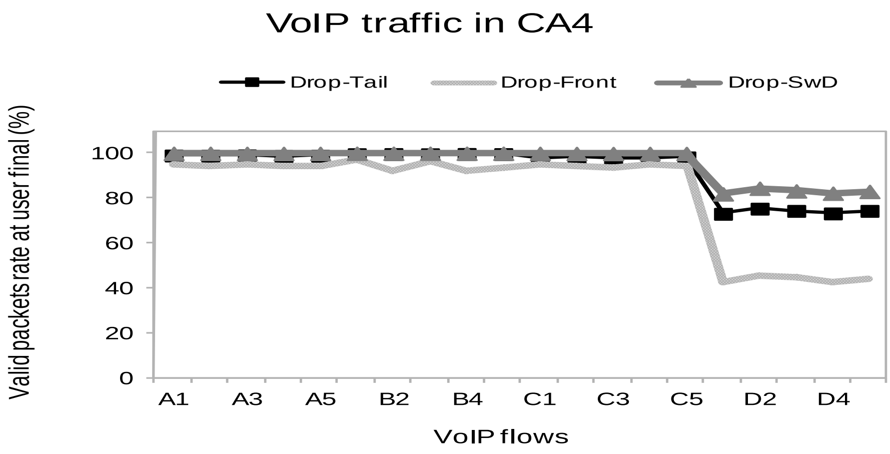


b)

Figure 12. VoIP packets loss rate at AQM REM for cases CA3 and CA4.



a)

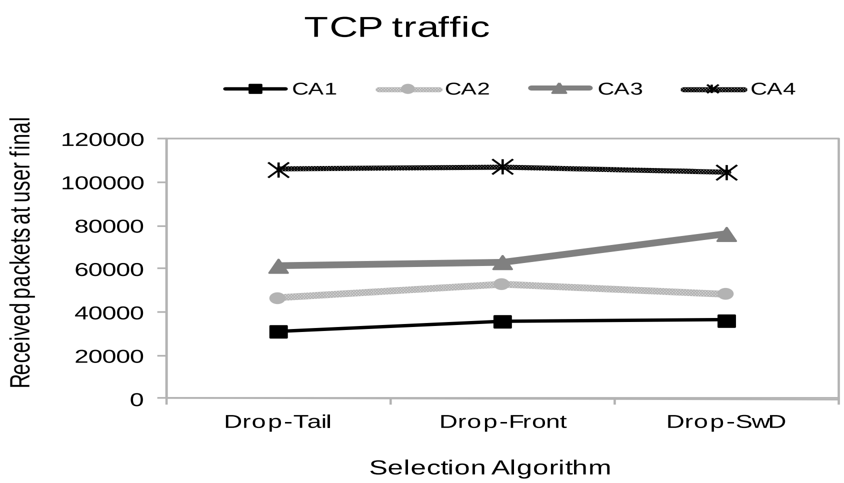


b)

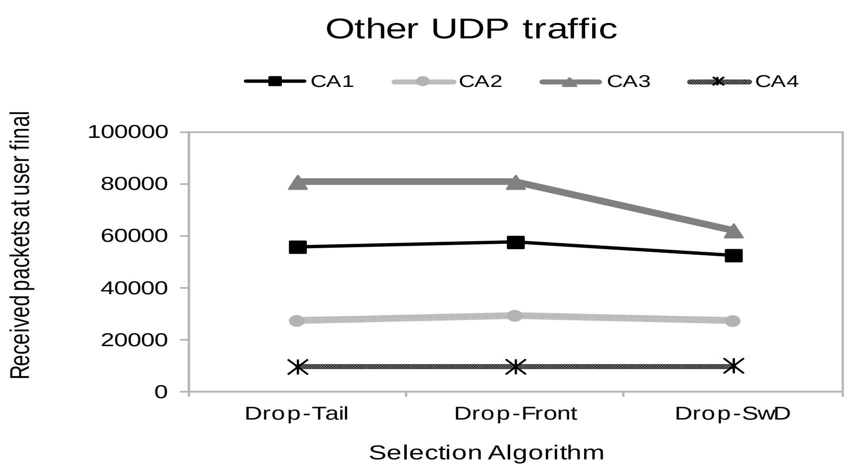
Figure 13. Valid VoIP packets rate at user final for cases CA3 and CA4.

6.2 TCP Traffic and Other UDP Non-real-time Traffic

With the goal of measuring the impact of *Drop-SwD* in the other two types of traffic (TCP and other UDP non-*real-time* traffic), the number of received packets for each type of traffic have been investigated. The results, which are presented in Fig. 14 and Fig. 15, indicate that *Drop-SwD* increases the amount of received valid VoIP packets without drastically affecting the arrival rate of the other flows. We should note that in case CA3, where the other UDP non-*real-time* traffic is dominant, the algorithm increases the dropping rate of this type of traffic for both AQM. This behavior promotes fairness for dealing with the TCP and VoIP traffic.

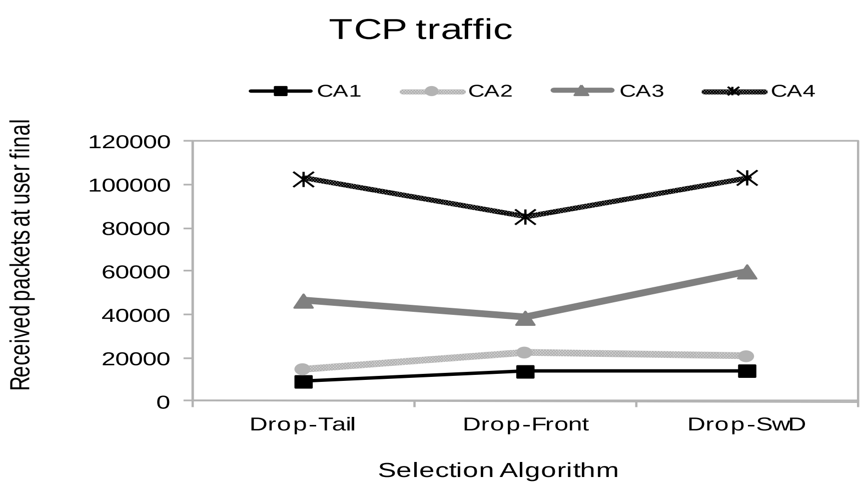


a)



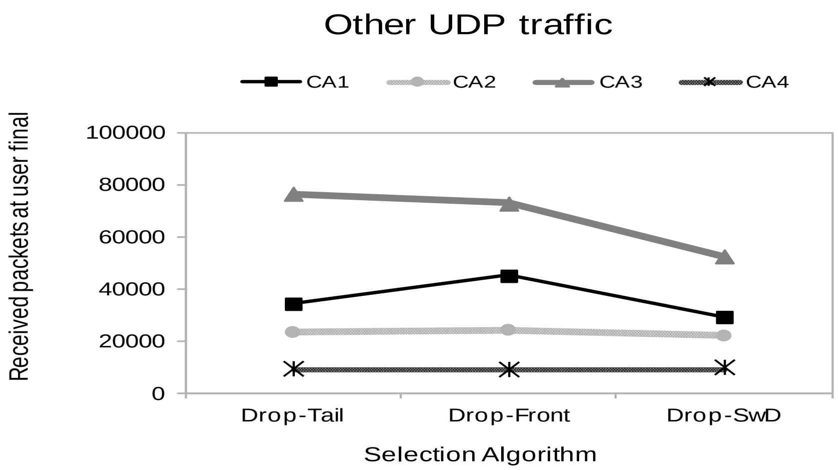
b)

Figure 14. Received packets at user final in RED scheme.



a)

Figure 15. Received TCP packets at user final in REM scheme.



b)

Figure 15. Received Other UDP packets at user final in REM scheme.

6.3 End-to-end Delay and Queuing Delay

Based on the effects of the delay introduced in the VoIP service (in relation to packet loss), we have performed an analysis on the queuing -time intervals in the router and the average end-to-end delay experienced by valid VoIP packets. The results that are generated with the different selection schemes in AQM RED and AQM REM are summarized in Table 3 and Table 4, respectively.

Table 3. Delay of Valid Audio Packets for AQM RED.

	Victim selection algorithms used in AQM RED								
	Drop-Tail			***Drop-Front***			***Drop-SwD***		
Case	Queuig delay (ms)	End-to-end delay (ms)	Variance (ms)	Queuing delay (ms)	End-to-end delay (ms)	Variance (ms)	Queuing delay (ms)	End-to-end delay (ms)	Variance (ms)
CA1	82	142	1.8	68	139	3.4	73	135	1.9
CA2	81	141	1.8	72	140	1.7	75	137	1.8
CA3	61	135	3.3	65	139	3.4	45	135	5.8
CA4	50	146	3.4	67	153	2.1	42	142	3.9

Table 4. Delay of Valid Audio Packets for AQM REM.

	Victim selection algorithms used in AQM REM								
	Drop-Tail			***Drop-Front***			***Drop-SwD***		
Case	Queuig delay (ms)	End-to-end delay (ms)	Variance (ms)	Queuing delay (ms)	End-to-end delay (ms)	Variance (ms)	Queuing delay (ms)	End-to-end delay (ms)	Variance (ms)
CA1	50	119	4.1	62	136	3.8	30	117	6.2
CA2	56	125	3.8	68	130	1.9	44	120	4.5
CA3	42	123	5.0	36	119	5.1	29	122	6.3
CA4	41	140	3.9	57	149	2.8	37	138	4.1

Note that for cases CA3 and CA4, *Drop-SwD* reduced the queuing time experienced in the AQM RED compared with the other two victim selection algorithms. However, for the REM scheme, *Drop-SwD* is capable of reducing the delay in all four cases, thereby increasing the number of valid packets, as was observed in Fig. 11 and Fig. 13.

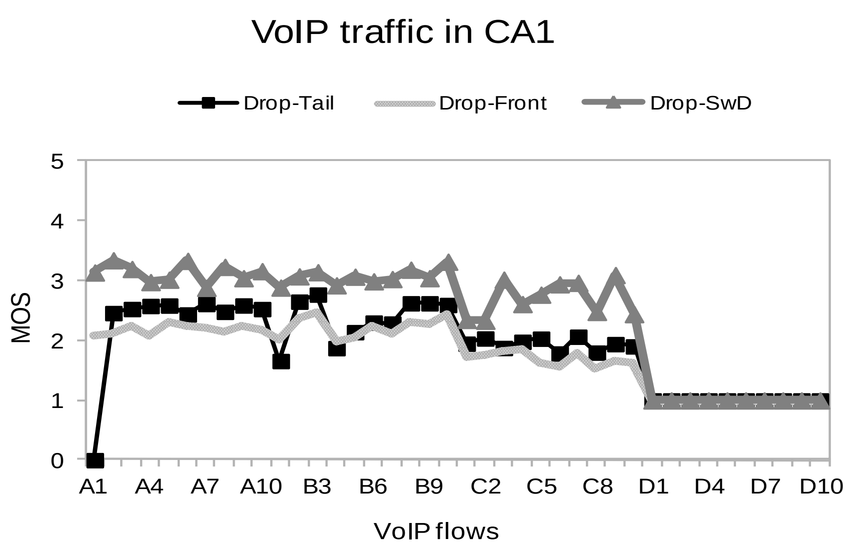
7. QoE Impact

For the case of general multimedia communication and for VoIP applications in particular, providing excellent performance in terms of QoS is worthless if it is not associated with excellent QoE.

7.1 E-Model Quality Evaluation

We have examined the performance of *Drop-SwD* in terms of perceptual quality in the end user. To this end, we have characterized the quality in terms of the R-factor using the E-model and adopting the configuration proposed in [22] and the ITU-T Recommendation G.107 [23]. The E-model predicts the subjective effect of combinations of impairments (such as noise, echo and signal loss) on the effect of individual impairment, the result is given in R-factor punctuation which varies from 0 (worst case) to 100 (excellent). To provide more readable subjective evaluations, the R-factor can be mapped to the MOS score. This factor is turned into a MOS score scale through the equations described in [22].

Fig. 16 and Fig. 17 show the MOS scores obtained for the different flows using *Drop-SwD* applied in RED and REM AQM schemes in different cases (CA1, CA2, CA3 and CA4). For the sake of comparison, we have also presented the scores obtained with the *Drop-Tail* and *Drop-Front* algorithms.



a)

Figure 16. E-MODEL based MOS evaluation of VoIP packet by flow in RED.

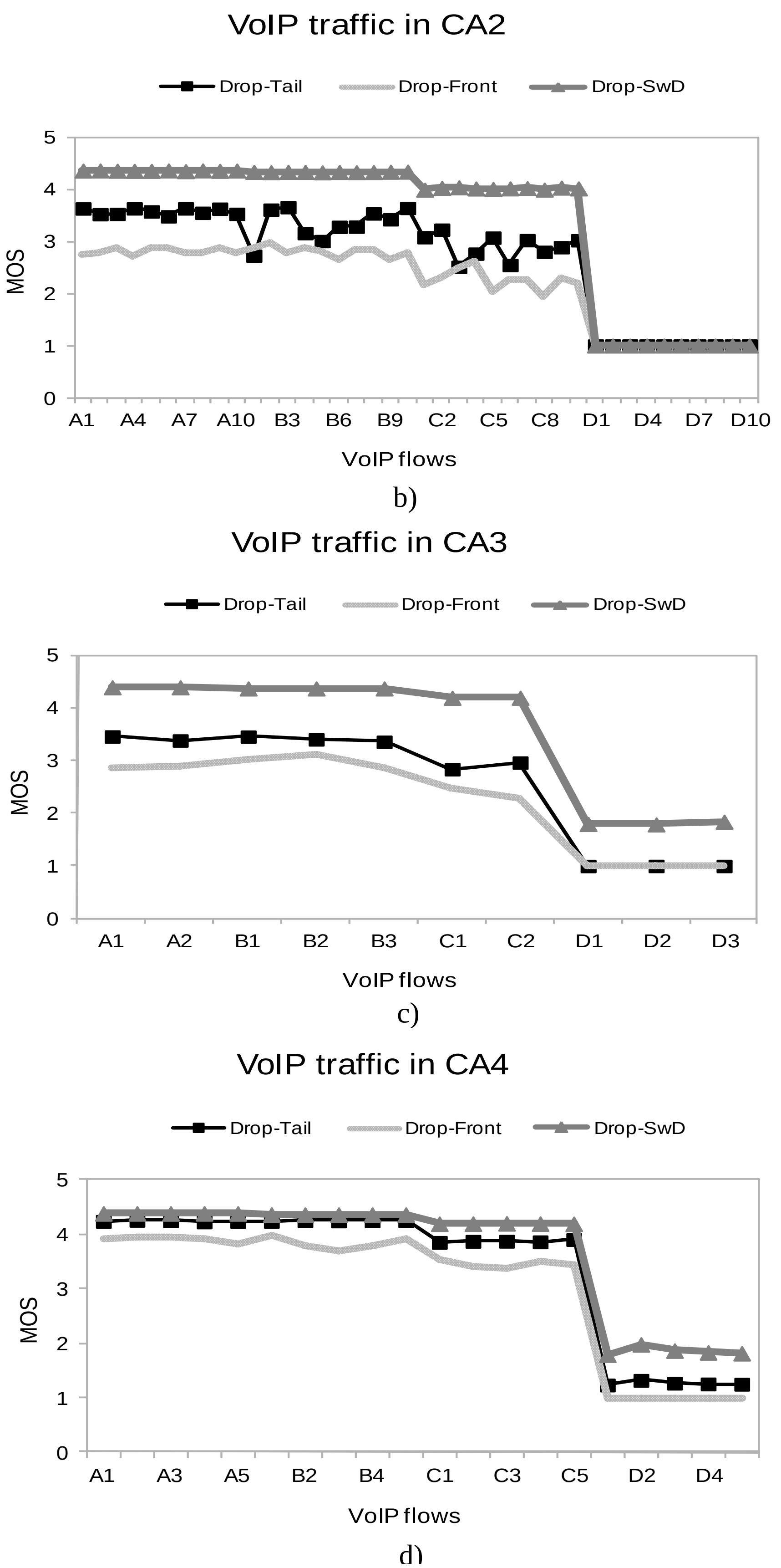
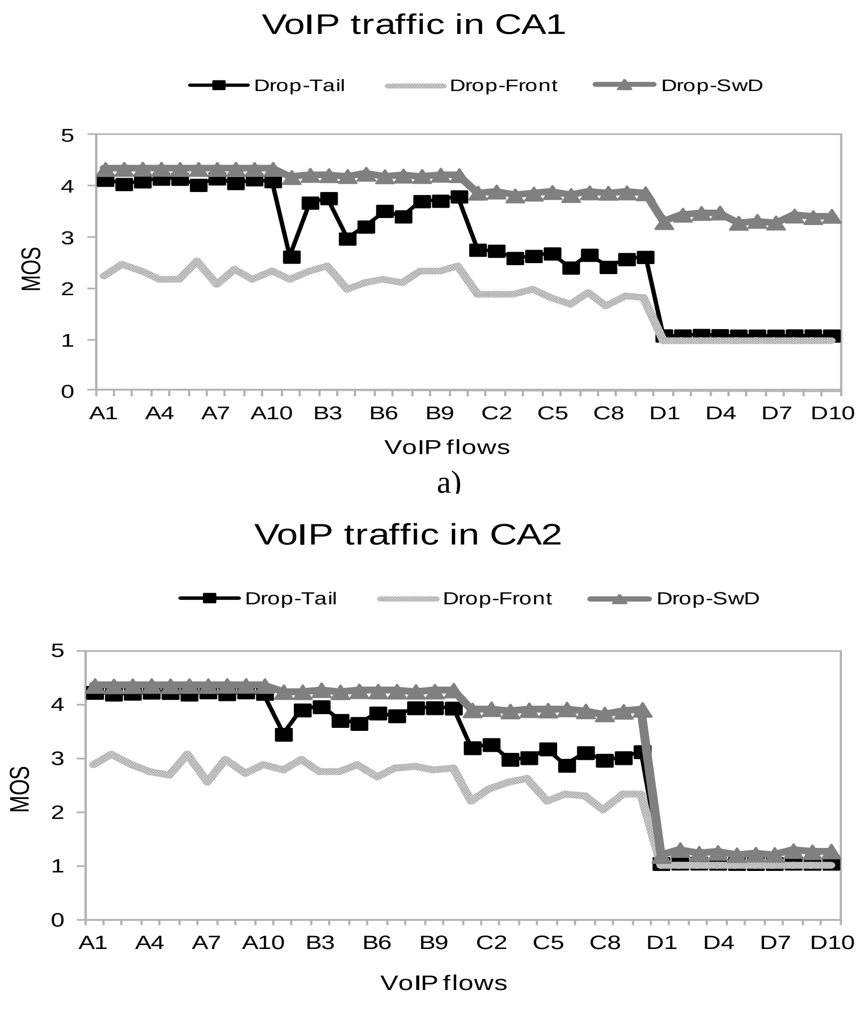


Figure 16. E-MODEL based MOS evaluation of VoIP packet by flow in RED.

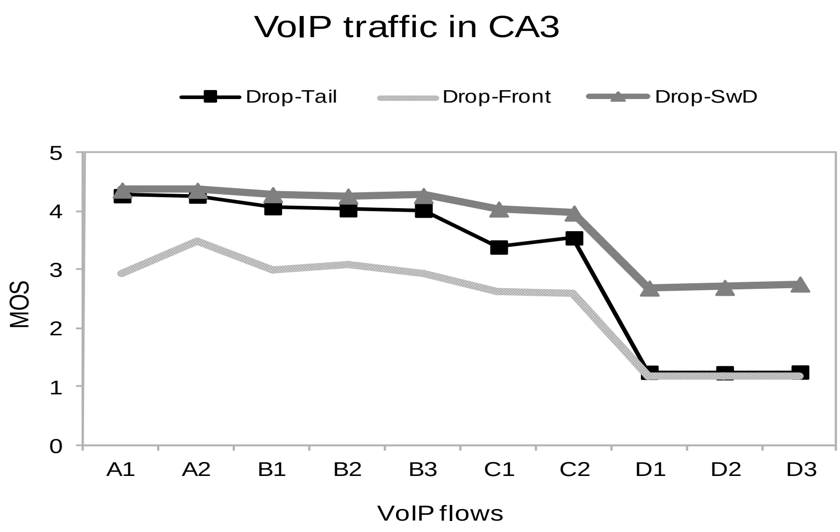
For the different cases, the flows that are less affected by the end-to-end delay (types A, B and C) experience the lowest queue drop rates in the AQM queue and in the receptor with *Drop-SwD* according to the QoS analysis of the results. This level of packet loss is directly related to the impact on the level of quality perceived by the user, as can be observed in Fig. 16 and Fig. 17. The MOS scores obtained with *Drop-SwD* for the typical flows A, B and C are superior to the scores of the other tested methods. In the cases CA2 and CA3 (see Fig. 16b and Fig. 16c), the including *Drop-SwD* procedure in RED increases the transmission quality from a medium level to a very high level. For case CA1, the including selective algorithm in REM increases the quality from a poor level to a medium level (see Fig. 17a).

We must note that the flows that are most affected by the end-to-end delay (type D in Table 1) have the lowest MOS scores. However, *Drop-SwD* provides these flows a similar or greater quality relative to the other schemes. The presented results indicate that *Drop-SwD* has a lower impact on quality degradation due to switching from one victim selection algorithm to another (when the VoIP sources dominate) and because it conserves valid packets in the queue while penalizing invalid packets.

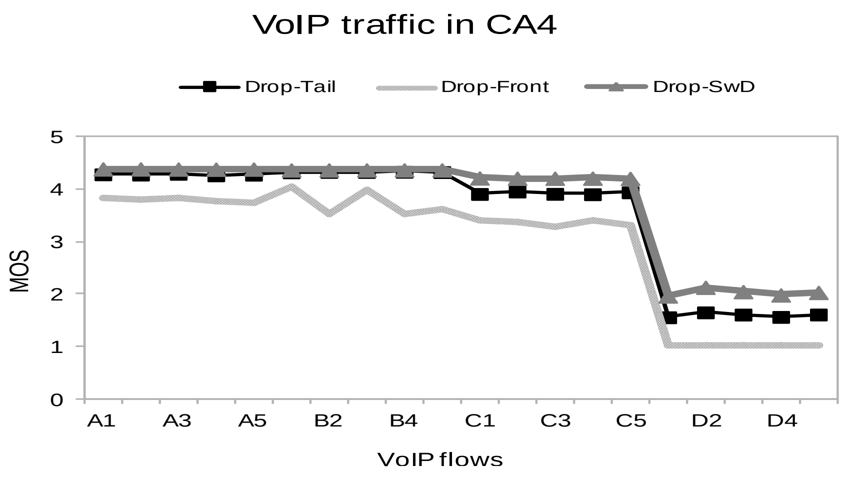


b)

Figure 17. E-MODEL based MOS evaluation of VoIP packet by flow in REM.



c)



d)

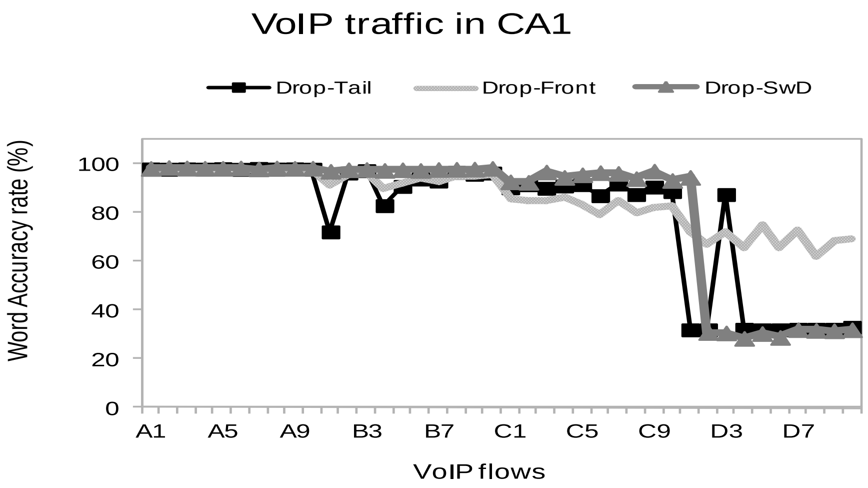
Figure 17. E-MODEL based MOS evaluation of VoIP packet by flow in REM.

7.2 Automatic Voice Recognition

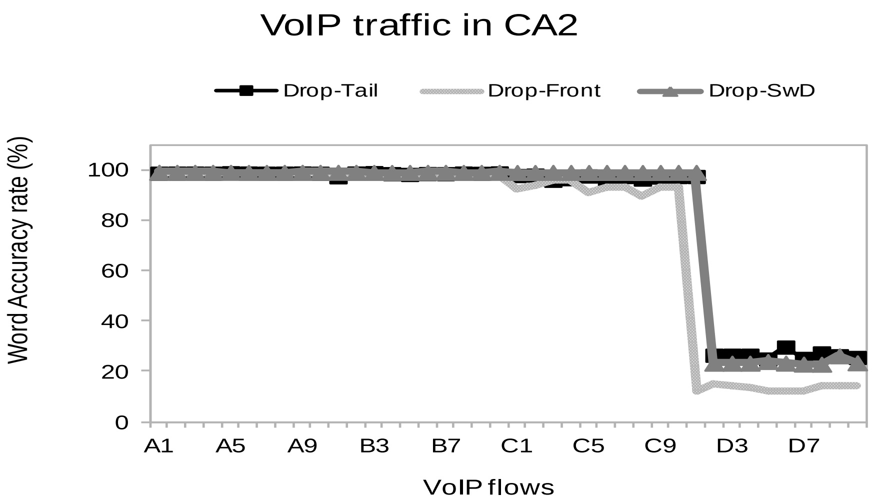
In an attempt to provide a complementary evaluation mechanism to MOS, we have considered a scheme based on automatic voice recognition to analyze the intelligibility of the flows (measured as a function of word rates or phrase precision) [24] [25]. This scheme is relevant to applications involving the transmission of multimedia information. With this estimation, higher levels of intelligibility, clarity and resulting quality will be associated with a higher score or success rate, which is obtained in an automatic recognizer located in the destination node.

7.2.1 Word Accuracy Rate

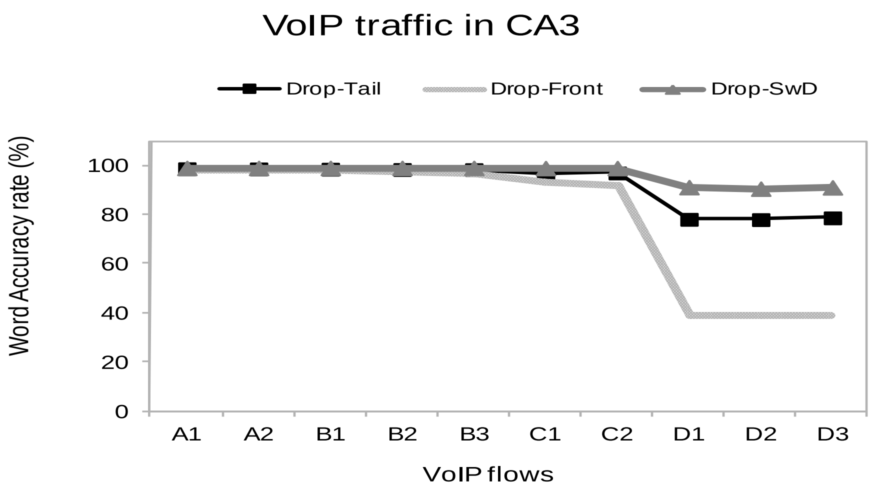
During the first evaluation phase in the recognizer, we measured the word accuracy rates. Fig. 18 presents the results obtained with RED. For cases CA1 and CA2 (see Fig. 18a and 18b), all of the mechanisms provide similar levels of intelligibility in terms of word precision. However, in cases CA3 and CA4 (see Fig 18c and 18d), *Drop-SwD* has a greater impact on the less favored flows in the experienced end-to-end delay with respect to the other two selection algorithms. For example, in case CA3, *Drop-SwD* contributed to the increase in the percentage of intelligibility for the application type D flows (with respect to *Drop-Tail*) in terms of word precision, from 78.25% to 90.96%. In case CA4, this value increased from 65.03% to 85.35%.



a)

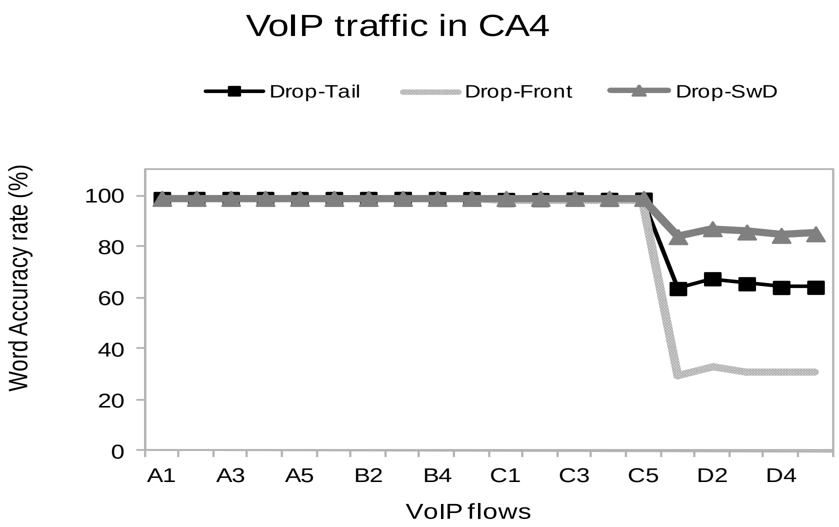


b)



c)

Figure 18. Word Accuracy rates of VoIP packet by flow in RED scheme.

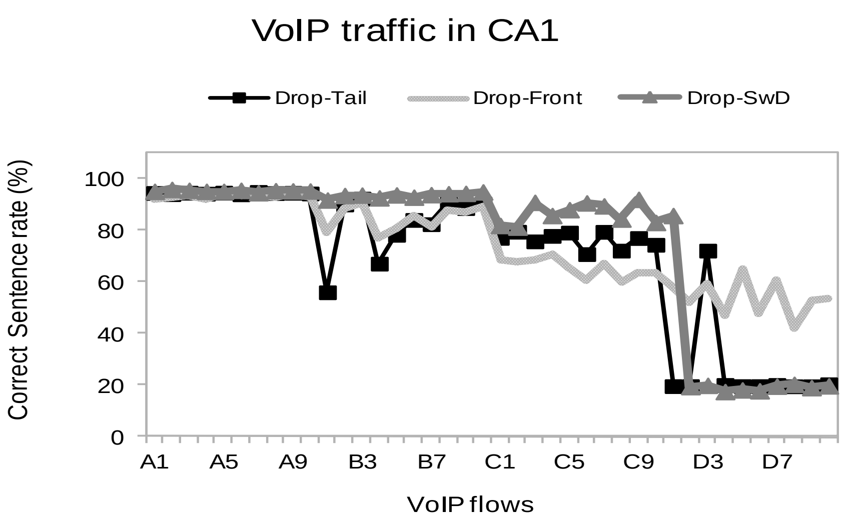


d)

Figure 18. Word Accuracy rates of VoIP packet by flow in RED scheme.

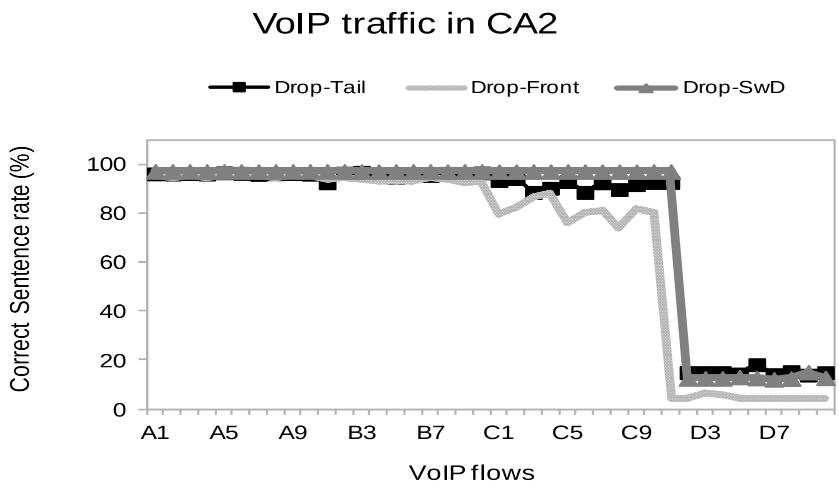
7.2.2 Correct Sentence Rate

In addition to the word accuracy rate, a complementary intelligibility measurement, the correct sentence rate, was also considered. A sentence is considered to be correctly recognized when there are no insertions or substitutions in it. In this context, Fig. 19 shows the correct sentence rate obtained when applying the proposed strategy in RED for the previously considered cases. As observed, *Drop-SwD* achieves an equal to or greater intelligibility than *Drop-Tail* and *Drop-Front*. It can be seen that there exist a correspondence between the results of Fig. 19 and the results in terms of word accuracy rate. The benefit of the selective algorithm in cases CA3 and CA4 (see Fig 19c and 19d) is more significant for the flow with the worst quality, which corresponds to voice-generated flows derived from source node that are further away. The selective dropping of packets was more efficient relative to the other two dropping schemes.

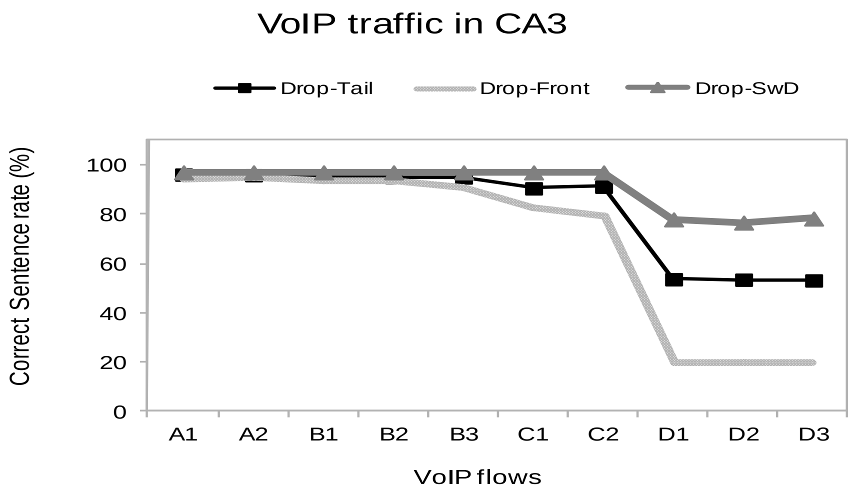


a)

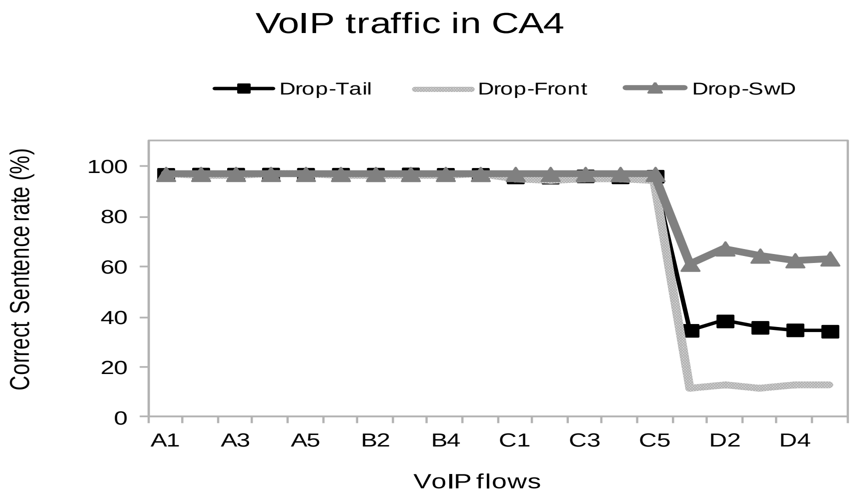
Figure 19. Correct Sentence rates of VoIP packet by flow in RED scheme.



b)



c)



d)

Figure 19. Correct Sentence rates of VoIP packet by flow in RED scheme.

8. Conclusions and Future Work

In this paper, we propose and evaluate *Drop-SwD*, which is a simple victim selection mechanism for AQM schemes and aims to provide a certain quality of experience to VoIP flows. Contrary to other proposals, *Drop-SwD* considers the interactivity demands of the real-time service for VoIP packets. Therefore, whenever the AQM determines that a packet has to be dropped, *Drop-SwD* identifies the traffic class with the highest router memory consumption (responsible for the congestion) and penalizes it. If VoIP traffic is responsible for the congestion, *Drop-SwD* switches to another traditional strategy for victim selection mechanism. However, if it is necessary choose a VoIP packet to drop; *Drop-SwD* selects the audio packet that will have the least impact on the perceived QoS. Therefore, it selects VoIP packets with longer end-to-end estimated delays because these packets will be dropped (at the receiver) due to the interactive nature of VoIP flows. As a consequence, our scheme enables better use of the network resources by forwarding only valid packets.

The objective evaluation reported indicates that *Drop-SwD* can increase the number of valid voice packets (never exceeds the maximum alive time permitted) received at user final. In addition, the conducted subjective evaluation shows that *Drop-SwD* obtains better quality than the other victim selection mechanisms applied in the evaluated AQM schemes (REM and RED).

As future work, we plan to extend *Drop-SwD* to consider other real-time media flows such as video from videoconferencing applications. In that sense, the use of TCP and HTTP as transport protocols for streaming media with time constraints [26] will be also considered to adapt the *Drop-SwD* algorithm to cope with that class of TCP traffic.

Acknowledgement

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References

[1] Braden R., Clark D. and Shenker S., “Integrated Services in the Internet Architecture: an Overview”. RFC 1633. 1994. Available at: http://www.ietf.org/rfc/rfc1633.txt (last accessed November 1, 2011)

[2] Grossman D., “New Terminology and Clarifications for Diffserv”. RFC 3260. 2002. Available at: http://www.rfc-editor.org/rfc/rfc3260.txt (last accessed November 1, 2011)

[3] Braden B., Clark D., Crowcroft J., Davie B., Davie B., Estrin D., Floyd S., Jacobson V., Minshall G., Partridge C., Peterson L., Ramakrishnan K., Shenker S., Wroclawski J. and Zhang L., “Recommendations on Queue Management and Congestion Avoidance in the Internet”. RFC 2309. 1998. Available at: http://tools.ietf.org/html/rfc2309 (last accessed November 1, 2011)

[4] Erika P. Alvarez-Flores, Juan J. Ramos-Munoz, Pablo Ameigeiras and Juan M. Lopez-Soler, “Selective Packet dropping for VoIP and TCP flows”. Telecommunication Systems. Vol. 46, Number 1. Pp. 1-16. 2011. http://dx.doi.org/10.1007/s11235-009-9252-z

[5] Floyd S. and Jacobson V., “Random Early Detection Gateways for Congestion Avoidance”, IEEE/ACM Transactions on Networking. Volume 1, Issue 4, August 1993.

http://dx.doi.org/10.1109/90.251892

[6] Joo C., Hong J. and Bahk S., “Assuring Drop Probability for Delay-Insensitive Traffic in a Differentiated Service Network”, Consumer Communications and Networking Conference (CCNC 2005). April 2005. http://dx.doi.org/10.1109/CCNC.2005.1405226

[7] Technical Specification from Cisco, "Distributed Weighted Random Early Detection". Available at: http://www.cisco.com/univercd/cc/td/doclproducsoftware/ios/wred.pdf

[8] Bodin U., Schelen O. and Pink S., "Load-tolerant differentiation with active queue management," ACM SIGCOMM Computer Communication Review, vol. 30, Issue 3, pp. 4-16, July 2000. http://dx.doi.org/10.1145/382179.382180

[9] Cao Z., Wang Z. and Zegura E., “Rainbow fair queuing: fair bandwidth sharing without per-flow state”, Nineteenth Annual Joint Conference of the IEEE Computer and Communications Societies. IEEE INFOCOM 2000. pp. 922–931. 2000. http://dx.doi.org/10.1109/INFCOM.2000.832267

[10] Chung J. and Claypool M., “Dynamic-CBT and ChIPS-Router Support for improved Multimedia Performance on the Internet”, In Proceedings of the ACM Multimedia Conference. 2000. http://dx.doi.org/10.1145/354384.354492

[11] Yamaguchi T. and Takahashi Y., “A queue management algorithm for fair bandwidth allocation”, Computer Communications, Volume 30, Issue 9, 30 June 2007. pp 2048-2059. 2007. http://dx.doi.org/10.1016/j.comcom.2007.04.002

[12] Chan S.P., Kok C.W. and Wong A. K., “Multimedia Streaming Gateway with Jitter Detection”, IEEE Transactions on Multimedia. Vol. 7, No.3. pp 585-592. 2005. http://dx.doi.org/10.1109/TMM.2005.843338

[13] Phirke V., Claypool M. and Kinicki R., “Traffic Sensitive Active Queue Management for Improved Multimedia Streaming”. Lecture Notes in Computer Science, 2003, Volume 2601/2003, 551-566, http://dx.doi.org/10.1007/3-540-36480-3_40.

[14] Mamatas L. and Tsaoussidis V., “Less Impact Better Service (LIBS). A service paradigm for Internet telephony”. Annals of Telecommunications. Vol. 65, No. 7-8, pp 447-459. 2010. http://dx.doi.org/10.1007/s12243-010-0175-1

[15] Dimitriou S. and Tsaoussidis V., “Promoting effective service differentiation with Size-oriented Queue Management”. Computer Networks, Vol. 54, No. 18, pp. 3360-3372. 2010. http://dx.doi.org/10.1016/j.comnet.2010.07.002

[16] Perkins C. and Schierl T., “Rapid Synchronisation of RTP Flows”. RFC 6051. 2010. Available at: http://www.rfc-editor.org/rfc/rfc6051.txt (last accessed November 1, 2011)

[17] Ns2, Network Simulator. Available at: http://www.isi.edu/nsnam/ns/ (last accessed November 1, 2011)

[18] Athuraliya S., Low S.H., Li V.H. and Qinghe Y., "REM: active queue management", IEEE Network, Vol.15, No.3, pp.48-53, May 2001. http://dx.doi.org/10.1109/65.923940

[19] International Telecommunication Union Standardization “Pulse code modulation (PCM) of voice frequencies. Recommendation G.711”. 1998.

[20] Iacovoni G., Morsa S., Parisi D. and Pierotti D., “Broadband FRA scenario and traffic modeling”. IST-1999-11571 EMBRACE D 22. 15th. April 2002,

[21] Liu S.G., Wang P.J. and Qu L.J., “Modeling and Simulation of self similar data

traffic", In Proceedings of the Fourth International Conference on Machine Learning and Cybernetic. pp. 3921-3925. 2005. http://dx.doi.org/10.1109/ICMLC.2005.1527623

[22] Cole R.G. and Rocenbluth J.H., "Voice over IP Performance Monitoring", ACM SIGCOMM Computer Communication review, Vol. 31, Issue 2. pp 269-275. April 2001. http://dx.doi.org/10.1145/505666.505669

[23] International Telecommunication Union Standardization. "The E-Model, a computational model for use in transmission planning". Recommendation G.107. 2005.

[24] Jiang W. and Schulzrinne H., "Speech recognition performance as an effective perceived quality predictor", Tenth IEEE International Workshop on Quality of Service (2002). pp. 269-275. 2002. http://dx.doi.org/10.1109/IWQoS.2002.1006595

[25] Juan J. Ramos-Muñoz, Ángel M. Gómez and Juan M. Lopez-Soler, "Intelligibility evaluation of a VoIP Multi-flow Block Interleaver", Lecture Notes in Computer Science, pp. 200-205. 2009. http://dx.doi.org/10.1007/978-3-642-00972-3_18

[26] Pantos R., "HTTP Live Streaming", IETF Internet-Draft, September 30, 2011.

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