A SIMULATION ANALYSIS OF AGGREGATION STRATEGIES IN A WF^{2}Q+ SCHEDULERS NETWORK

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***Abstract* -- The paper presents an analysis in a DiffServ network scenario of the achievable QoS (Quality of Service) performance when different aggregation strategies between video and voice traffic flows are considered. Each network node of the analyzed DiffServ scenario is represented by a Worst-Case Fair Weighted Fair Queueing scheduler with a shaper obtained using a Shaped Starting Potential Fair Queueing. The system of each node, referred in literature as WF^{2}Q+ scheduler, permits to guarantee the isolation among the different PHB (Per Hop Behavior) service classes, maintaining the multiplexing gain. The parameters setting of the WF^{2}Q+ scheduler is also discussed. In particular, the necessary network resources, estimated by the WF^{2}Q+ parameters setting obtained considering the aggregation of traffic sources belonging to the same service class, are compared with those estimated on a per flow basis. The higher gain achievable using the first approach with respect to the second one, is also qualitatively highlighted. The simulation results, presented in the paper, evidence the possible problems that can be raised when voice traffic is merged with video service traffic. As a consequence, the paper results suggest to consider in different service class queues the two kinds of traffic.**

Keywords -- WF^{2}Q+ scheduler, LBAP traffic characterization, DiffServ architecture, voice model, QoS parameters

1. Introduction

A key challenge of the current telecommunication age is represented by the developing of new architecture models for IP networks in order to satisfy the recent QoS requirements of innovative IP-based services (e.g. IP Telephony and videoconferencing).

At present, the ISP (Internet Service Provider) often provide the same service level independently from the traffic generated by their clients. Taking into account the transformation of Internet to a commercial infrastructure, it is possible to understand the need to provide differentiated services to users with widely different service requirements.

In this framework, the DiffServ approach is the most promising for implementing scalable service differentiation in IP networks. The scalability is achieved by considering the aggregate traffic flows and conditioning the ingoing traffic at the edge of the network. Aggregation obviously decreases the complexity of traffic control in the core network, but it produces some unwelcome effects, such as “lock-out” or “full-queues” phenomena, which contribute to increase end-to-end delay and jitter of the traffic flow of a single service. These two phenomena take place respectively when few flows monopolize queue space preventing other connections from getting in the queue and when it is not possible to maintain the queues non-full. Hence, the effects of the aggregation mechanisms on the QoS parameters of the different aggregated flows need to be further analyzed. The first works in this field have highlighted relevant concepts to support traffic aggregation [1], however a still open issue is what kind of aggregation strategies is better to carry out. To this aim we investigate traffic aggregation strategies because there is still no clear position on what is the better configuration (standardization organisms say anything regarding this matter). On the other hand recent publication [2] suggests to divide network traffic in only two service classes (e.g. real-time and non real-time) but it seems, from our point of view, a little bit restrictive with respect to different traffic features. In the paper, we analyze the impact of the aggregation of real-time video and voice traffic in the same service class (hence, in the same queue in a per-service queueing system) on the experimented QoS parameters of the different flows. The QoS concept used in this work is to be identified with the whole set of properties which characterize network traffic (e.g. in terms of resource availability, end-to-end delay, delay jitter, throughput and loss probability). The results obtained in this scenario are then compared with those obtained considering real time video and voice as separate traffic flows.

The DiffServ architecture [3] is a good starting point but it is useless if there is no teletraffic engineering background able to provide the needed differentiation. Therefore we should take into account also scheduling disciplines and their dimensioning, in order to understand if they may affect results (changing scheduling discipline change the way the flows are treated). Hence, we have firstly chosen to use one of the best work-conserving scheduling algorithms (WF2Q+) instead of a non work-conserving one used in other analysis [4]. The choice derives from the assumption that the best the scheduling algorithm is (keeping acceptable its complexity) the better treatment a flow receives in terms of low end-to-end delay and jitter. Moreover, the parameters setting of the considered scheduling discipline has been analyzed as described is in Section 4.

In the analysis, we use as traffic characterization approach, the LBAP (Linear Bounded Arrival Processes) theory [5]. Furthermore, we investigate the effects on LBAP characterization of the multiplexing of the video traffic, instead of taking into account the simple sum of the traffic descriptors obtained with the single source. By means of simulation analysis we evaluate if the multiplexing gain derived from characterization of aggregated traffic does not affect the QoS parameters.

The rest of the paper is organized as follows. In Section 2 we present the simulation scenario while in Section 3 we describe the voice source model, the video and data traffic taken into account in the simulation analysis. In Section 5, the results are discussed while Section 6 summarizes the main results presented in the paper.

2. Simulation Scenario

Our simulation scenario mainly reflects the topology of a DiffServ domain of an IP network. The simulation scenario is implemented using the OPNET Modeler vers. 6.0.L, a powerful CAMAD (Computer Aided Modeling and Design) tool used in modeling communication systems and in analyzing network performance. The considered scenario is shown in Fig. 2.1.

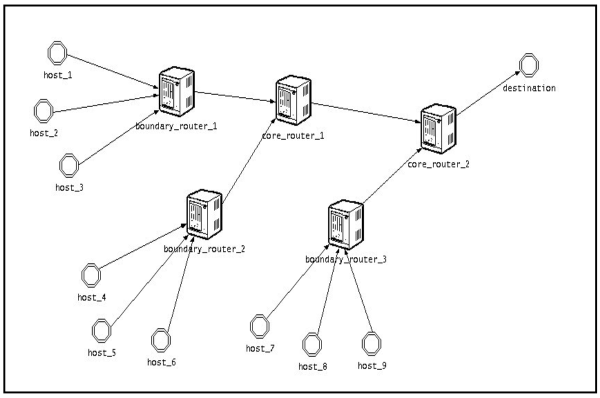

Fig 2.1: Simulation scenario

The network model is represented by edge and core routers, each one having a work conserving scheduler that permit to realize the isolation of the entering flows, based on performance guarantees. The scheduling discipline is a Worst-Case Fair Weighted Fair Queuing with the addition of a Shaper, obtained using a Shaped Starting Potential Fair Queuing (SSPFQ), denoted as WF^2Q+ [6]. The WF^2Q+ is a GPS (Generalized Processor Sharing) approximating service discipline with high fairness properties and relatively low implementation complexity. Moreover, in order to simulate a single *Diffserv* domain, we implement at the edge router the classifier and the marker necessary to associate each packet to the selected PHB. Based on this classification and marking, each packet receive the suitable forwarding treatment by the core routers. The traffic sources taken into account in the simulation scenario, are the most heterogeneous possible because we want to analyze the performance of a real network; it must integrate the carrying of video, voice and data traffic. Hence, describing the scenario shown in Fig. 2.1 in more details, every block named as host 1, 4 and 7 contains 15 voice sources, while every block named as host 3, 6 and 9 contains a video source. The remaining hosts contain “data” module, which simulate best-effort traffic.

The statistics we have collected concern the most significant QoS parameters of real-time services, i.e. end-to-end delay and jitter delay, which are evaluated considering the connection among the different sources and the destination node shown in Fig. 2.1.

3. Source models

In the simulations, we adopt a model only for the voice sources, while for the other kinds of traffic we consider actual traffic data.

The model used for the voice sources consists in an On-Off model, suggested by the typical behavior of a voice source with VAD (Voice Activity Detection): it is active or inactive depending on the talker is speaking or silent. Assuming that no compression is applied to voice signal, during active periods the source transmits at the constant bit rate of v=64 Kbps (this corresponds to a standard PCM codec with VAD). In-depth analyses of this traffic source, shown in literature, have emphasized that the distribution of active and inactive periods lengths can be approximated by an exponential function [7], with mean values respectively equal to T_{on}=350 msec and T_{off}=650 msec. The packet size is 64 bytes, and considering the bit rate and the header overhead (40 bytes taking into account the RTP/UDP/IP header) the source generates one packet every 3 msec.

Video flow	Mean_rate (Mbps)	Peak_rate (Mbps)
GOLDFINGER	0.584	5.87
ASTERIX	0.537	3.54
SIMPSONS	0.446	5.77

Table 3.1 – Statistical parameters of considered video sources

The traffic data used for the video sources are described in [8], where also their statistical analysis is presented. They have been obtained collecting the output of an MPEG-1 encoder loaded by different sequences of movies half an hour long. Some relevant statistical parameters of the traffic data used, named Goldfinger, Asterix and Simpsons, are summarized in Table 3.1.

The video packets are produced at application level, dividing the number of bytes produced by the encoder in the frame period, T=1/24 sec, in consecutive packets of size equal to 1500 bytes (in this case an MTU, Maximum Transfer Unit, of 1500 is supposed). Moreover, in the simulation we consider the 40 byte of overhead, related to the UDP/RTP/IP header, assuming that every packet transports 1460 byte of the traffic data registered at the output of the encoder.

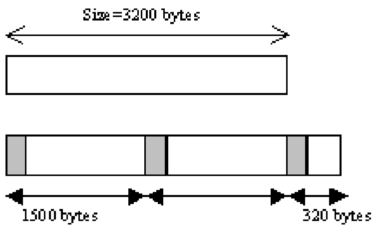

Fig 3.1 - An example of packet fragmentation at application level

The time interval between the generation of the consecutive packet in each frame period, has been considered deterministic and equal to T/N, where N is the number of packets needed to transport all the bytes produced by the encoder during a frame period. As an example, Fig. 3.1 presents a case where 3200 bytes are necessary for the encoding of a frame; at application level we divide the frame in three packets, which are sent with time interval equal to T/3 sec.

The Best Effort sources have been obtained considering the traffic data acquired at the Faculty of Engineering of the University of Pisa. In particular, we consider the traffic exchanges by the Faculty of Engineering with the external world (essentially other University sites and Internet) by means of an ATM network at 155 Mbps [9]. The peak rate of the considered traffic is equal to 11 Mbps, while a mean rate of only 400 Kbps has been observed; the high peak-to-mean ratio is a clear evidence of the high burstiness of the data traffic. During the data acquisition both the arrival time and the size of each packet have been registered. Hence, in this case the packet generation process to use in the simulations is directly obtained from the traffic data.

4. Parameters Setting

In order to set the scheduler parameters, we first characterize the traffic sources by mean of the LBAP approach. The traffic characterization of the single source is obtained by the parameters (*b, ρ*) where *b* is indicated as bucket size and *ρ* as token rate. The physical interpretation of the LBAP parameters can be understood, considering that the number of bytes produced by a single source in a time interval of (0,τ), A(τ), is upper bounded by

$$A(\tau) \le b + \rho\tau \qquad \forall\ \tau>0$$

The results presented in [10] permits to have an upper bound for the end-to-end delay experimented by the traffic when it traverses through a Latency Rate scheduler network, as that considered in our simulation scenario (i.e. WF^2Q+); estimation of WF^2Q+ latency term is described in [11]. In particular, the delay introduced by a single node to a packet belonging to the *i-th* flow, characterized by the LBAP parameters (b_i, ρ_i) is upper bounded by (in the following we suppose that for the *i-th* flow a service rate equal to ρ_i is allocated)

$$D \le \frac{b_i}{\rho_i} + \Theta_i$$

where Θ_i represents the latency of the scheduler, defined as $\Theta_i = \dfrac{L_{i,\max}}{\rho_i} + \dfrac{L_{\max}}{C}$. $L_{i,max}$ and L_{max} respectively represent the maximum packet size of the *i-th* flow and of the global traffic arriving to the scheduler, while ρ_i and C, are the service rate allocated to the *i-th* flow and the global output service rate respectively. Extending the analysis to a network of *K* WF^2Q+ schedulers, the end-to-end delay experimented by a single packet belonging to the *i-th* flow is upper bounded by

$$D_i \le \frac{b_i}{\rho_i} + \sum_{j=1}^{K} \Theta_i^j ,$$

where Θ_i^j indicates the latency of the *j-th* node evaluated for the *i-th* flow.

The end-to-end delay bound has been obtained considering the worst-case analysis, which is more conservative with respect to experimented end-to-end delay.

Furthermore, as shown in [12], also the LBAP traffic characterization is conservative with respect to the statistical modeling approaches. These two considerations leads us to assume that the maximum end-to-end delay of the *i-th* flow can be upper bounded simply by

$$D_i \le \frac{b_i}{\rho_i} .$$

This hypothesis permits to establish the buffer size and the guaranteed rate to set in the scheduler, simply evaluating the LBAP curve of the *i-th* flow.

The simulation results that will be presented in Section 5 point out the goodness of our assumption, showing the very conservative nature of the worst-case analysis.

Considering the above hypothesis, the procedure used to set the scheduler parameters consists in evaluating the LBAP curve and in finding the point where this curve intersect the straight line $b_i = \rho_i D_i$, where D_i represents the maximum delay fixed for the considered source.

Figure 4.1 shows the presented approach in the case of the video sources. In particular, in the figure we can observe the LBAP curves for three different video sources, and the curve related to the traffic obtained aggregating these sources. Moreover, assuming a maximum end-to-end delay of 200 msec, we can observe the relate straight line and the intersection points with each LBAP curves that, as described above, give the couple *(ρ_i, b_i)* to consider in the setting of scheduler parameters.

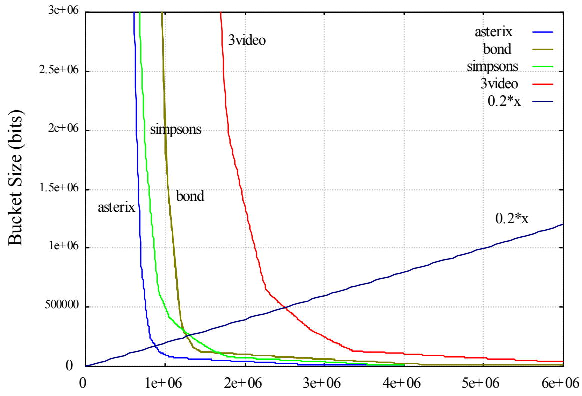

Fig. 4.1 - Video characterization

The characterization results obtained considering the considered three video flows (named Goldfinger, Asterix and Simpson), the aggregate of 15 voice sources (corresponding to a host in the simulation scenario) and the data traffic are reported in Table 4.1. In the table, the column Dmax indicated the maximum end-to-end delay analytically obtained from the estimation of scheduler parameters.

Traffic flow	**Rate** (ρ) (Mbps)	**Buffer** (b) (Kbit)	**Dmax** (ρ/b) (msec)
GOLDFINGER	1.25	250	200
ASTERIX	0.83	160	193
SIMPSONS	1.27	260	205
15 VOICE sources	1.10	30	27
Data	0.40	400	1000

Table 4.1 - Traffic characterization of considered sources

In the setting of the scheduler parameters, we can choice to set a service rate equal to the sum of ρ obtained for the three sources, and a buffer size equal to the sum of *b* obtained for each source. Considering this approach and the results obtained with the procedure presented above, the total service rate to allocate for the video services and the related buffer size are equal to 3.35 Mbps and 670 Kbits respectively. The other approach consists in the estimation of the parameters directly from the LBAP curve of the multiplexed traffic. In this case, it is expected to obtain a multiplexing gain in the setting of the resources to guarantee to the video service. In particular, considering the same upper bound of the end-to-end delay, we need to allocate a service rate of 2.5 Mbps and a buffer size of 500 Kbits. Then, in terms of buffer size we observe a gain of 170 Kbits (corresponding to a reduction of about 25%), while in terms of service rate a gain of 850 Kbps (25%) is achieved. Considering the service rate evaluated for each kind of traffic source, it is possible to set the scheduler parameter assuming a utilization factor of the link equal to 0.9. In more details, we evaluate the sum of ρ_i, ρ_{tot}, and fix the rate of the output link equal to $\rho_{tot}/0.9$. Hence, for boundary and core routers, the parameters are set as summarized in Table 4.2.

	Output Service Rate (Mbps)	Buffer Size, B (Kbit)
Boundary Routers	3.06	B_{voice}=30
		B_{data}=500
		B_{video}=250
Core router 1	6.12	B_{voice}=60
		B_{data}=1.000
		B_{video}=500
Core router 2	9.18	B_{voice}=90
		B_{data}=1.500
		B_{video}=750

Table 4.2 – Parameters of the routers

The buffer size is given for each traffic class, i.e. voice, video and data. When considering the scenario related to the aggregation of voice and video flows, the buffer size for this aggregated class of traffic has been set equal to $B_{voice}+B_{video}$.

5. Simulation results

The simulation analysis is mainly focused in the evaluation of the impact of different aggregation strategies on the QOS parameters. The low scalabity of the IntServ network architecture, suggests for the IP Telephony scenario to study a DiffServ core network architecture. Hence, it is expected that in each node of an IP Telephony core network, a per-class queueing is implemented and an appropriate scheduling algorithm guarantees the QoS requested by the real time sources.

Two relevant problems arise in this framework. The first concerns the choice of an appropriate scheduling algorithm and of a procedure for the setting of their parameters. The second issue is related to the aggregation strategies to adopt. Hence, in this Section we present the results that give insights on these problems.

We consider two different aggregation strategies: in a first one we have carried video and voice together in a premium class and data in a best effort class (in the figures the related curves are indicated with label containing the word "1 link"); in a second one we have supposed to carry video traffic in a separate queue from voice traffic (curves labeled with the noun containing the word "2 link").

Fig 5.1 presents the complementary probability of the end-to-end delay experimented by the voice packets when the two different strategies are considered and only one video source is activated (in this case the setting of simulation parameters have been changed according to the dimensioning procedure presented in the previous paragraph, in order to take into account the inactivity of the others video sources). Fig. 5.2 presents the same curves obtained when all three video sources are active.

As first analysis of the simulation results, it is possible to note that the adopted scheduling algorithm, i.e. WF^2Q+, and the procedure for its parameters setting permit to guarantee the target QoS for the voice sources if the video and voice traffic flows aren't merged in a single queue. In particular, the curves labeled as "1 Link" either in Fig. 5.1 and in Fig. 5.2 clearly show that the maximum delay observed during the simulation is under 10 msec, which is lower than the fixed delay of 27 msec, considered in the setting of the scheduler parameters.

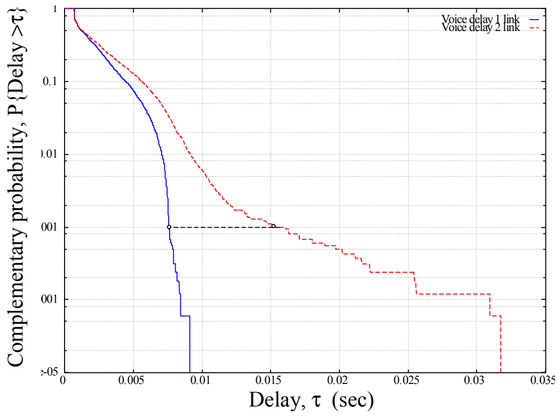

Fig 5.1 - Complementary Probability of Voice Delay: P{delay > τ} - One Active Video Source

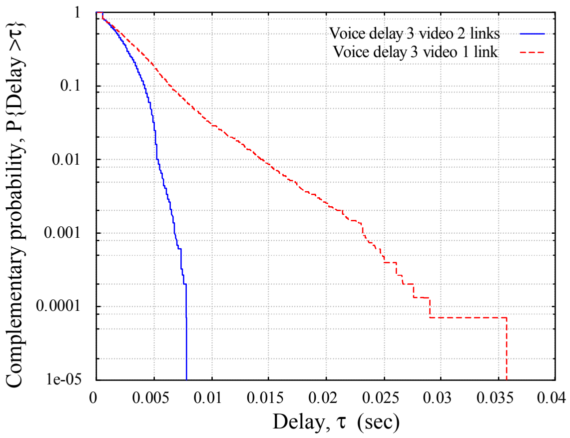

Fig 5.2 - Complementary Probability of Voice Delay: P{delay > τ} - Three Active Video Source

Both Fig. 5.1 and Fig. 5.2 show the degradation in terms of end-to-end delays of voice traffic when the video flows is merged in the same queue with the voice traffic. Indeed, in the Fig. 5.1, we can note that in correspondence of a probability P=0.001 a delay of 7 msec is observed in the first case (curve "1 Link"), which is lower than the 15 msec registered in the second case.

Furthermore, this degradation is amplified when the number of video sources is increased. Indeed, in Fig. 5.2 the maximum end-to-end delay registered for voice traffic is unvaried with respect to the previous case, i.e. about 7 msec, while it is increased to 23 msec, when all real-time flows are aggregated in the same queue.

Hence, in this second case the worsening of the delay parameter of voice service is due to the increase of the number of bursty traffic multiplexed with the voice sources. Hence, we can suppose that if there is more requested bandwidth the need for network resources increase in a non-linear way when considering a wrong strategy of aggregation, in this case the real time application may be damaged seriously.

On the other hand, the video performance take benefits from the aggregation, showing a little delay improvement that can be related to the multiplexing with the voice (the related figures are not reported for sake of simplicity).

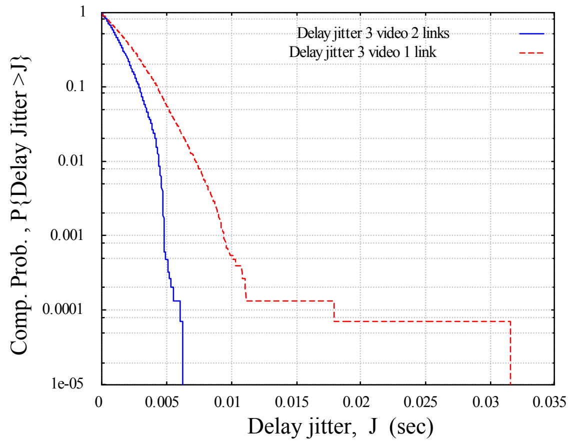

Fig 5.3 - Complementary Probability of Voice Jitter Delay: P{ delay jitter > τ} - Three Active Video Source

The different performance observed with the two considered aggregation strategies, can be related to "lock-out" phenomenon, which plays a decisive role in deteriorating voice performance. Indeed, when the video sources are merged with the voice traffic, the first monopolize the queue space obstructing the second from receiving the desired service level. The "lock-out" phenomenon can be avoided using different queues for video and voice traffic, while, at the same time, the choice of appropriate scheduling disciplines can guarantees an adequate multiplexing gain.

Finally, we observed that the best effort traffic (used as background traffic) is not affected by the fusion of the two service classes because the total bandwidth share (video + voice) is unchanged.

The same worsening of performance can be observed when we consider the jitter parameter, as shown in Fig. 5.3, which plots the complementary probability estimated for this statistic (the results are related to the multiplexing of three video sources).

6. Conclusion

The main goal of the paper is the evaluation of different traffic aggregation strategies for voice and video services in a *Diffserv* environment. In this framework, the simulation analysis presented in the paper highlights that the wrong aggregation of traffic flows with different statistical features, such as video and voice traffic, may lead to performance worsening, which should be avoided especially in providing IP-based business services, such as IP Telephony.

On the other hand, the simulation results emphasize that with an adequate isolation between video and voice traffic flows and an appropriate dimensioning of network resources, it is possible to provide real-time services. In particular, the considered WF^2Q+ schedulers network and the proposed procedure for setting the related parameters, permit to achieve the target QoS. Furthermore, analyzing the proposed procedure for the setting of schedulers parameters, based on the LBAP traffic characterization, it has been possible to highlight the multiplexing gain obtainable considering the LBAP characterization of aggregated traffic.

Finally, the simulation results have evidenced that although we have neglect the latency terms in the expression of the end-to-end delay reported in literature, the maximum delay experimented is lower than that analytically estimated (see the fourth column in Table 4.1). This result is a further evidence of the very conservative nature of the worst-case analysis.

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