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An applied research study for the provision of a QoS-oriented environment for voice and video services over satellite networks

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Abstract

The aim of this paper is to present a network solution to provide Quality of Service (QoS) guaranteed voice and video services over a satellite test-bed based on the TCP/IP protocol suite, oriented to distance learning. The scope of the solutions proposed is a small private satellite network where specific services (e.g. tele-education) are offered to the users.

The paper describes the main operating steps followed to perform the tests. The target service requirements are identified and the configuration of devices and protocols are described in detail. The Integrated Services approach is utilized along with three different scheduling mechanisms at IP layer. A proper routing protocol to support multicast is used.

The work is a part of a project called 'Integration of Multimedia Services over Heterogeneous Satellite Networks', which has been developed by the Italian National Consortium of Telecommunications (CNIT) and funded by the Italian Space Agency (ASI). The test-bed is composed of two remote sites, connected through a satellite link at 2 Mbits/s and by one site, where no satellite device is available, connected by using ISDN.

The results reported allow to test the application of bandwidth reservation strategies originally studied for a terrestrial environment and to measure the consequent performance improvement. The use of both subjective metrics as mean opinion score (MOS) and objective metrics, as audio/video packet loss and average bit rate, to perform the investigation, allow mapping the measure of P-QoS (Perceived-Quality of Service) with quantities uninfluenced by personal opinions. The configurations used represent the result of an applied research project and provide an operative solution, inclusive of technical details, to guarantee high quality of service for audio/video applications.

Keywords: Satellite networks; TCP/IP; Quality of Service

1. Introduction

The development of telecommunication networks [1] has allowed a widespread use of applications as tele-education, tele-working, tele-meeting, where people joining the event are not located in the same room, but in remote sites. The mentioned applications, along with many others, are mainly based on two components: audio and video, which require a precise level of quality to be utilized. At the same time, most applications are based on the TCP/IP protocol stack [2] that cannot guarantee Quality of Service (QoS), if used in standard version. Moreover, the need to have scalable network architecture, wide land coverage and multicast services suggests the use of a satellite network [3]. As a consequence, it is necessary to match three different components, each of them introducing specific problems and deserving a special attention: Quality of Service (QoS)—guaranteed audio and video service, utilization of the TCP/IP stack, satellite environment.

Concerning the support of QoS over IP satellite networks, there are different actions to take, which depend on the protocol layer. If each layer might be completely redesigned, ignoring the products already available in the market, the literature would offer a great amount of possibilities. For instance, at the application layer, flow control algorithms that adapt the bit rate entering the network might improve the performance of the overall communication [4–6]. At the transport layer, the performance of the TCP may be greatly improved by tuning algorithms and parameters [7–11], and by adapting them to the satellite environment.

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In this paper, as only real time audio and video services are considered, the UDP protocol is used at the transport layer. As a consequence, the modifications of the TCP, so effective in TCP-based services as web access, file downloading and tele-control, may be ignored. On the other hand, the action at the network layer (IP) is topical in this context. The strategies proposed in the literature may be classified into two main models: Integrated Services [12] and Differentiated Services [13]. The former allows assigning a certain bandwidth to a specific traffic flow. The mechanism uses a signaling protocol called ReSerVation Protocol (RSVP) [14] to transmit the bandwidth needs. The strategy is aimed at transforming a best-effort network into a QoS-guaranteed network (e.g. ATM) by imposing a virtual planning over the physical link. Differentiated Services use the characteristics of the IP packet (e.g. the Type of Service field) to provide a fixed degree of service to a group of traffic flows. The Differentiated Services approach is very attractive and, for its scalability, it is the ideal candidate to be the QoS-oriented solution in the Internet of the future. On the other hand, also the possibility to provide QoS to a single traffic flow is attractive. Especially in a small private network, where each customer deserves attention, as in tele-education systems, the chance to distinguish each user by using the source/destination IP address and the source/destination TCP–UDP port is really useful. As a consequence, the Integrated Services approach has been chosen for the support of QoS in the test-bed.

This work is part of a wider project aimed at investigating novel solutions for multimedia services over satellite networks. The project, called 'Integration of Multimedia Services over Heterogeneous Satellite Networks', carried on by CNIT and funded by ASI, is divided into three work packages: transmission layer, network layer, transport layer. The last two parts are strictly integrated, and actually, the solutions at the transport layer have been implemented together with the proposals (presented in this paper) to provide an overall distance-learning service based on real time audio–video communication and web navigation. The protocol architecture has been presented along with few important results, suited for a tutorial paper, in Ref. [15], which, even if it is performed within the same experimental environment, does not contain detailed description of the configurations used and does not distinguish between the audio and video flow composing the video-conference service under test. The present paper, which is an extended version of the paper in Ref. [16], focuses on the audio–video service and shows the configuration of the devices used in detail. The effect of different IP packet scheduling schemes is analyzed by varying the video bit rate and by adding some traffic disturbance. The Quality of Service Perceived by the users (P-QoS) has been measured by using a mean opinion score (MOS) method. The evaluations vary from value 1 (poor) to value 5 (very good). The MOS values, along with measures of objective metrics, as audio/video packet loss and average bit rate, have been reported. All the results have been obtained through real work sessions. Mbone tools (SDR [17] for multicast sessions announcement, VIC [18] for video and RAT [19] for audio, see Ref. [20] for a detailed description of the overall software tool) have been used to transmit video and voice. No dynamic flow control at application level is applied.

The test-bed is composed of two sites, located in Genoa and Florence, connected through a satellite link at 2 Mbits/s and of a single site, located in Pisa, where no satellite device is available, connected by using ISDN. The network dimension is limited but it represents a real test-bed to experiment and implement network solutions.

It is important to say that similar solutions could also be applied in terrestrial networks not including satellite portions. The results obtained allow testing strategies, as RSVP and Fair Queuing, formerly studied for cabled environments, and to measure the difference in the performance by using them. The use, for example, of RSVP has a cost, both in computing and human resources, and it is important to know precisely which is the benefit provided to the final user. In particular, it is necessary to establish, if and how much the use of RSVP is useful and in which application context. The contemporary use of subjective (MOS) and objective metrics allows performing a mapping among them. The control mechanisms in the literature are often based on quantities independent of the personal feelings and ignore the practical effect on the final user. On one hand, this methodology permits to separate the design from a specific context (and it is recommendable for network devices as switches, routers, gateways), but, on the other hand, does not consider the final user, who is fundamental in many end-to-end services. The results reported allow to check which is the real effect on the users of an objective performance improvement (a packet loss reduction, for instance), if it is really meaningful for the service provision, so that the right decision may be taken for the implementation of QoS-oriented control algorithms. The overall study described in the paper reflects an applied research project and contains operative solutions (obtained through the integration of tools already available in the market) for QoS-oriented environments dedicated to audio and video communication.

The paper is structured as follows. Section 2 contains the description of the experimental test-bed. Section 3 identifies the main characteristics of the Integrated Services approach. The system configurations used in the test-bed are summarized in Section 4. Section 5 contains the results and Section 6, the conclusions.

2. Experimental test-bed

Three CNIT remote sites (Genoa, Florence and Pisa) have been connected through a heterogeneous wide area network (Fig. 1), where there are communication links with different characteristics in terms of propagation delay and BER. In

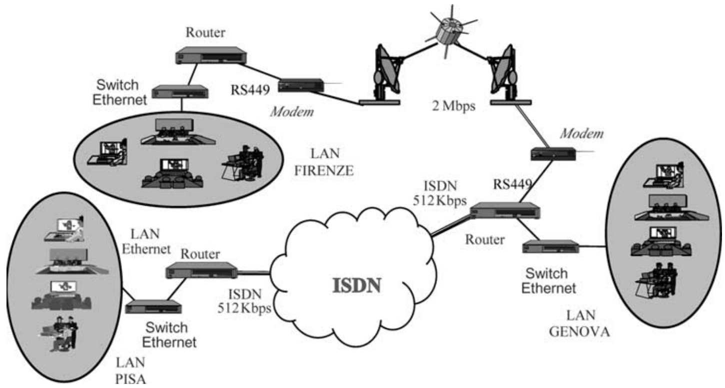

Fig. 1. Experimental framework.

particular, a satellite link at 2 Mbit/s has been used for the interconnection of a LAN located in Genoa with a LAN situated at Florence, whereas a LAN in Pisa has been interconnected to the satellite network aggregating four BRI ISDN channels.

Each satellite site has the following devices:

- a satellite station, composed of a 1.8 m antenna, operating in the Ka-band (20–30 GHz) and aiming at the Italsat II satellite, a BB (Base Band)–RF (Radio Frequency) up/down converter and a base-band modem
- an IP Router, acting as a gateway between the terrestrial and the satellite networks and equipped with an RS449 Serial Interface, a 10BaseT Ethernet Interface and four ISDN BRI interfaces
- some PCs, equipped with a video capture card, performing H.261 encoding at 15 frames per second and connected to the LAN through an Ethernet card at 10/100 Mbits/s.

On the other hand, each terrestrial site requires

- an IP Router, equipped with a 10BaseT Ethernet Interface and four ISDN BRI interfaces;
- some PCs, as for the satellite sites.

3. The integrated services approach

In terms of QoS requirements, real-time applications, such as audio and video, are very sensitive to end-to-end delay, delay jitter and excessive packet loss. Therefore, real-time applications do not perform well in traditional IP networks that only provide a best-effort packet delivery service with no guarantee as it concerns timeliness or actual packet delivery. New architectural components must be added to the TCP/IP protocol architecture to allocate network resources for QoS sensitive applications and to make high-quality networked multimedia applications a reality. The Internet Engineering Task Force (IETF) has recently proposed many service models to meet the demand for QoS. In particular, the most relevant ones, are the Integrated Services (IS)/Resource Reservation Protocol (RSVP) [12] and the Differentiated Services (DS) [13] models. In the CNIT-ASI Intranet, the Integrated Services model has been chosen and implemented, because scalability is not an issue and the perfect isolation of the data flows is a strict constraint.

The Integrated Services network architecture is characterized by resource reservation. It is implemented by four components: the signaling protocol (RSVP), the admission control routine, the classifier and the packet scheduler [21].

Before the data are transmitted, real-time applications must first set up paths and reserve resources. RSVP [14] is the signaling protocol designed by the IETF Integrated Services Working Group to allow applications to communicate their QoS requirements. It enables RSVP capable applications, running on an end-system host to send resource reservation requests to the destination system and to specify the QoS parameters for a data flow. Two service classes have been proposed in addition to the best-effort service: the controlled-load [22] and guaranteed [23] services. They, respectively, provide a 'reliable and enhanced best-effort service' and 'a fixed delay bound service'.

Within the CNIT–ASI Intranet, taking advantage of some router features, video and audio applications have not been modified to support RSVP signaling protocol: actually, the routers directly connected to the LANs of the transmitting hosts have been configured as if they were receiving signaling messages from them. Once an application has sent its QoS requests, each router, running the admission control routine, must determine if there are enough network resources to accept the reservation request coming from the new flow without damaging the service level guaranteed to the flows already accepted. If the QoS request coming from a new flow is accepted, the reservation instance in the router assigns the packet classifier and the packet scheduler to reserve the requested QoS for the flow. The classifying action can be performed based on any parameter contained in the packet header, such as the source and destination addresses, the protocol, the session identifiers (port/sockets). The packet scheduler is responsible for the manner in which the queued

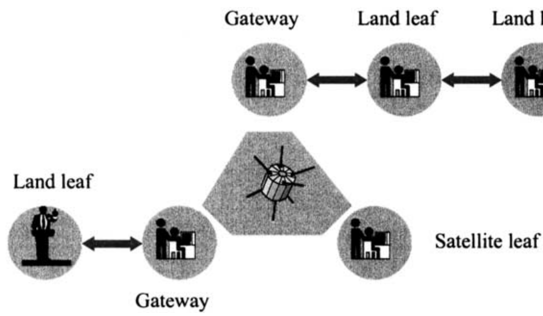

Fig. 2. Sites classification.

Table 1
Application parameters

Application	Parameter	Value
VIC	Motion compensation quality	10
	Video encoding	H.261
	Frame rate	15 fps
	Image format	CIF (352 × 288)
	Bit rate	128, 256, 384, 512 kbits/s
RAT	Audio encoding	PCM-64 kbits./s
	Resolution	16 bit
	Audio bandwidth	8 kHz

packets are selected for transmission on the link. RSVP can use a packet scheduler, such as the Weighted Fair Queuing (WFQ) to allocate buffer space, schedule packets, and guarantee bandwidth for reserved flows.

4. Implementation and configuration

This section highlights the main steps to follow for the design of a network architecture that satisfies the functional requirements described in the previous sections and for the configuration of commercially available devices.

4.1. Site classification

The high costs to sustain for the realization of a broadband satellite network make necessary to build a real, but down-scaled, satellite environment, where few stations play all the roles found in a larger satellite network. This can be obtained through the interconnection of three different types of sites, each of them belonging to one of the following classes:

- Satellite leaf site: it is a site characterized by a single, two-way connection to the satellite backbone.
- Land leaf site: it is a site directly connected through a terrestrial link (ISDN, ADSL, ATM, Frame Relay, etc.) to another 'Land leaf' or to a 'Gateway'.
- Gateway site: it is a site characterized by two or more different connections to other sites.

Fig. 2 shows a typical configuration of a down-scaled network, which can be used for distance learning applications. The classification of the sites is useful to group a set of configuration tasks and performance properties that are exhibited by all the sites in the same class.

4.2. Network requirements

Real-time voice and video services have the following network requirements.

4.2.1. Multicast support

Multicast is an efficient way of delivering one-to-many communications across an IP infrastructure. The use of unicast tranmission for this type of communication requires that each listener must make a separate connection to the server that is the source of the data. This results in tremendous load on the server and congestion across expensive WAN links as the number of listeners increases. With multicast, the server sends one stream to the network and a distribution tree forms. Interested listeners simply add a branch to the tree. Routers replicate packets at each branch in the tree.

In this way, no packets are ever duplicated in the network, and the server never has to send more than one stream of data. Therefore, in wide area network, each host, wishing to participate to a multicast session, must first inform the local multicast router of its desire to join the multicast group employing the Internet Group Management Protocol (IGMP) [24]. Then, the local router, using a multicast routing protocol interacts with the other routers to receive and to forward multicast packets. A complete multicast IP support has been provided in the experimental network activating IGMP version 2 at host level and PIM [24] Dense Mode within each network router.

4.2.2. QoS support

The Qos support is strictly related to the control of network resources and to their allocation based on application requirements. As packets belonging to various network flows are multiplexed together and queued for transmission at the output buffers associated with a link, the manner in which a router selects queued packets for transmission on each interface must be controlled. This can be accomplished by choosing and configuring a suitable scheduling discipline for each router interface.

The default FIFO service discipline allows a router to forward packets only in the same order in which they arrived at the output link queue. Other scheduling mechanisms, such as WFQ, ensure that queues do not starve for bandwidth, and that traffic gets predictable service.

WFQ can be used with or without a packet classifier. In the first case, it is possible to reserve a certain amount of bandwidth to a specific flow. In the second case, low-volume traffic streams, which comprise the majority of traffic, receive preferential service, transmitting their entire offered loads in a

Table 2
WFQ configuration parameters

Parameters	Ethernet interface	ISDN interface	Serial interface
Congestion discard threshold	64	64	64
Dynamic queues	256	256	256
Reservable queues	234	12	5

timely fashion. High-volume traffic streams share the remaining capacity proportionally between them.

In our test-bed, the Integrated Services has been chosen, as the reference model to support QoS and the Resource Reservation Protocol (RSVP) is the signaling protocol that allows applications running in hosts to reserve resources within network nodes. The full or partial insertion of the above mentioned components (classifier and scheduler) permits to create different scenarios where the applications can be tested and evaluated. In particular, the following operating conditions for testing the QoS support have been identified and used:

- FIFO Scheduling
- WFQ Scheduling, without specific classifier, also called Fair Queuing (FQ)
- WFQ Scheduling with RSVP.

4.2.3. Application requirements

The last element that influences the test-bed set-up is the application requirement definition. Distance Learning is a multimedia application basically composed of two services: multicast video-conferencing and multicast data dissemination. Mbone tools (VIC, RAT, and SDR) have been used to create video-conferencing sessions and to send audio and video, which are the flows of interest for this work. Table 1 summarizes the settings of the applications parameters used in the experiments.

4.2.4. Configuration of devices and protocols

To guarantee the quality of service requirements requested by audio and video flows, it is necessary to carefully plan the configuration of Weighted Fair Queuing and Resource Reservation Protocol within each router.

The configuration of WFQ on a router interface requires that the congestion threshold after which messages for high-bandwidth conversations are dropped and the number of dynamic and reservable queues be specified.

Table 2 shows, for each type of interface, the value assigned to the WFQ parameters to perform the tests.

To properly configure RSVP, it is necessary to take into account how much bandwidth should RSVP allow per-user application flow, how much bandwidth is available for RSVP and how much bandwidth should be excluded from RSVP.

In our scenario, two different RSVP reservations have been set on each router:

A *shared reservation* has been implemented for the multicast audio session. Since a single sender is normally active at any given time and, only sometimes, a limited number of senders are transmitting at the same time, a separate reservation for each sender is not required. Therefore, a single bandwidth reservation with a rate of 64 kbits/s has been applied to any sender within the set of the authorized participants to the session.

A *distinct reservation* should be implemented for the video session, because each sender emits a distinct data stream that requires admission and management in a queue. Each flow, therefore, would require a separate reservation per sender on each transmission facility it crosses. Nevertheless, in our scenario, a shared reservation has been implemented also for the video session. Each video flow has not been isolated and only the aggregated flow has been guaranteed. Taking into account that the default maximum bandwidth, which can be reserved on each router interface, is equal to the 75% of its available bandwidth and that 64 kbits/s has been reserved for the audio flows, a maximum guaranteed bandwidth

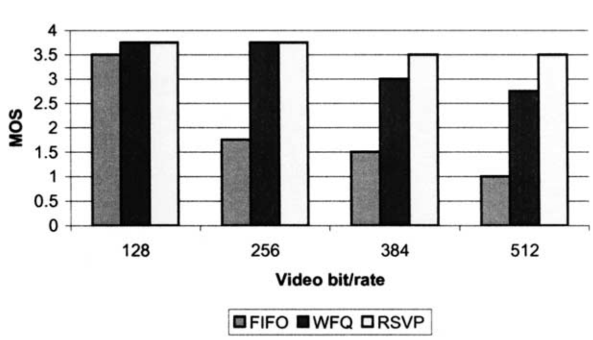

Fig. 3. Video-conference: P-QoS evaluation, no transfer.

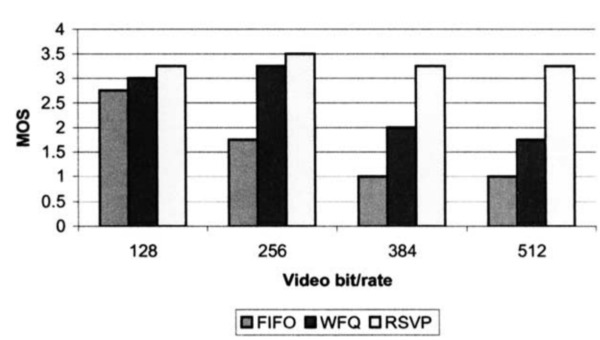

Fig. 4. Video: P-QoS evaluation, no transfer.

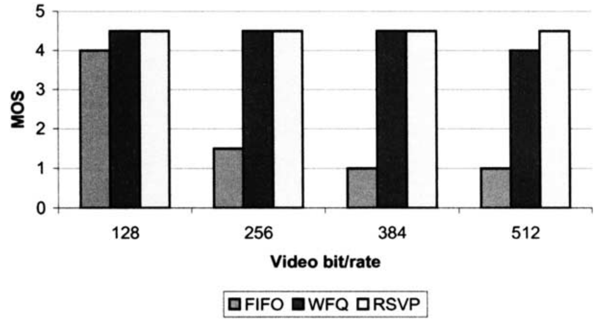

Fig. 5. Voice: P-QoS evaluation, no transfer.

reservation of 320 kbits/s has been installed. This bandwidth reservation is applied to any flow characterized by the destination port number and IP address of the multicast video session.

5. Results

The results presented are aimed at testing the performance of each component within the overall service (i.e. the distance learning oriented video-conference application). All the measures reported are an average of the results obtained in each of the sites connected.

The metrics chosen to evaluate the performance are

- Mean opinion score (MOS) value, which can vary from 1 (poor) to 5 (very good), to measure the P-QoS.
- Percentage of lost packets.
- Percentage Gain and Packet Loss Reduction (defined below).
- Average bit rate.

The quantities chosen, in the authors' view, should allow a complete investigation of the performance in all its aspects and match the aim of the paper.

The video-conference service under test is composed of two components: video and audio flow. The results are structured into three different groups of tests.

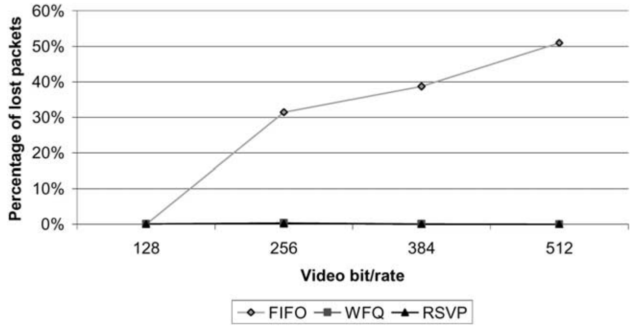

Fig. 6. Voice packet loss, no transfer.

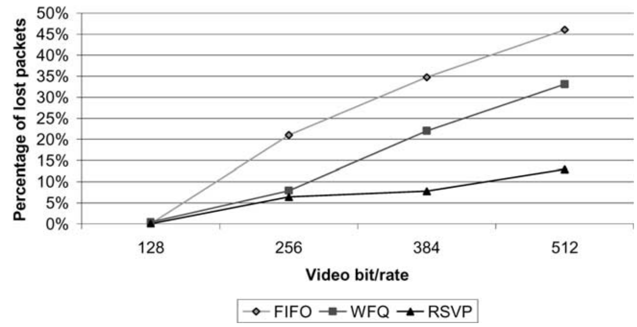

Fig. 7. Video packet loss, no transfer.

- A condition (identified as 'no transfer') where there is no disturbance traffic during the test sessions.
- A condition (identified as 'TCP transfer') where there is TCP-based disturbance file transfer during the test sessions.
- A condition (identified as 'UDP transfer') where there is an UDP-based disturbance 256 kbits/s video flow during the test sessions.

For each group of tests the following measures are reported:

- P-QoS for the overall video-conference service by using the FIFO Scheduling (identified as FIFO), the WFQ Scheduling, without any specific classifier (identified as WFQ), the WFQ Scheduling with RSVP (identified as RSVP), whose bandwidth reservations have been described in Section 4. The behavior of each scheduler has been evaluated with the video bit rate assuming four different values: 128, 256, 384, 512 kbits/s. The evaluation of the overall application had been already reported in Ref. [15]. Even if the performance analysis of this paper is focused on each single component, the overall measure has been maintained to avoid a limitation of the scope of the work and to allow investigating the effect of each single component and of the different mechanisms used on the overall service.
- P-QoS both for the video and the voice component of the service applying the FIFO, WFQ and RSVP scheduling and varying the values of the video bit rate (128, 256, 384, 512 kbits/s), as in the previous case.
- The percentage of lost packets both for the video and the voice component of the service applying the FIFO, WFQ and RSVP scheduling and varying the values of the video bit rate (128, 256, 384, 512 kbits/s).
- The gain in percentage and the packet loss reduction of all the WFQ and RSVP configurations by taking FIFO as a reference and of all the RSVP configurations if WFQ is used as a reference.

Two new measures are introduced for the test-bed situations where there is a disturbance. They allow, together the percentage of lost packets for the audio and video flow that are part of

Table 3
Gain in percentage, no transfer, FIFO as reference

Bit rate	Video-conference		Video		Audio	
	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)
128	7.1	7.1	9.1	18.2	12.5	12.5
256	114.3	114.3	85.7	100	200	200
384	100	133.3	100	225	350	350
512	175	250	75	225	300	350

the service (and, for this, should be protected against disturbances), to check the action of the different scheduling mechanisms in detail.

- The average bit rate of the TCP-based disturbance traffic.
- The packet loss of the UDP-based disturbance video flow.

The investigation described in this paper allows matching the following requirements:

- To map the P-QoS, i.e. the level of user perception, with objective metrics of quality (packet loss and average bit rate). The issue is of great importance. The P-QoS is often ignored during the design of QoS-oriented telecommunication systems. Objective quantities as loss, delay and jitter are the parameters on which the algorithms and the control applications are based, but not always outstanding performance improvements concerning objective metrics (e.g. packet loss reduction) correspond to P-QoS improvements of the same entity. So, if, on one hand, it is important to study control mechanisms based on objective quantities that do not depend on the single application and environment, on the other hand, it is also important (in particular for end-to-end systems) to study the real impact of the implemented strategies on the users who receive the service.
- To show that the schemes to reserve bandwidth, designed for terrestrial environments, as RSVP, are still effective in satellite environment. Even if, from the theoretical point of view, no problem should arise, the practical application is often different. In the case presented, heterogeneous networks, with different characteristics in terms of delay and BER (Bir Error Ratio), are traversed, including a satellite portion; many configurations have been necessary and an overall applied research project has been dedicated to tune all the details. The solutions provided are fully operative and include technical information.
- To show the difference among the different queuing strategies used (FIFO, WFQ, and RSVP) and to measure the performance improvement if a resource reservation mechanism is applied. The possibility of verifying the real improvement of a solution in a specific environment is not only interesting from a scientific point of view, but it allows finding out if and when a solution should be applied, considering the benefits of the solution and the cost introduced by it.

The measures reported in the paper, both the MOS and the objective quantities would require a higher degree of confidence. The number of users (about 10, globally) involved in the tests and the number of measures are not sufficient, on one hand, to smooth an opinion strongly different from the others and a strongly personal feeling, and on the other hand, to provide valid numerical values from a statistical point of view. Anyway, even if the measures cannot be considered completely reliable, they give a precise idea of the behavior of the metrics considered and of the macroscopic effect on the service.

Some definitions are necessary before presenting the results. The gain in percentage of the generic configuration x is defined as in formula (1), where MOS_x is the MOS value for the configuration x and $\mathrm{MOS}_{\mathrm{Ref}}$ is the MOS value for the reference configuration.

$$\%\mathrm{Gain} = \begin{cases} \dfrac{\mathrm{MOS}_x - \mathrm{MOS}_{\mathrm{Ref}}}{\mathrm{MOS}_{\mathrm{Ref}}} & \text{if } \mathrm{MOS}_x \geq \mathrm{MOS}_{\mathrm{Ref}} \\ 0 & \text{otherwise} \end{cases} \quad (1)$$

The possibility of $\mathrm{MOS}_x < \mathrm{MOS}_{\mathrm{Ref}}$, which should not be feasible because the reference is chosen as the configuration

Table 4
Gain, no transfer, WFQ as reference

Bit rate	Video-conference RSVP (%)	Video RSVP (%)	Audio RSVP (%)
128	0	8.3	0
256	0	7.7	0
384	16.7	62.5	0
512	27.3	85.7	12.5

Table 5
Packet loss reduction, no transfer, FIFO as reference

Bit rate	Voice packet loss		Video packet loss	
	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)
128	0	0	0	0
256	31.5	31.5	13.1	14.6
384	38.7	38.7	12.7	27
512	51	51	13	33.1

Table 6
Packet loss reduction, no transfer, WFQ as reference

Bit rate	Voice packet loss RSVP (%)	Video packet loss RSVP (%)
128	0	0
256	0	1.5
384	0	14.3
512	0	20.2

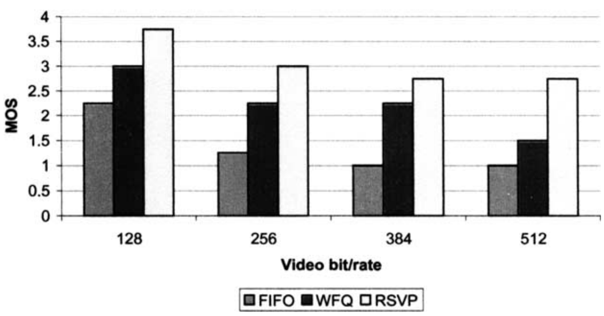

Fig. 8. Video-conference: P-QoS evaluation, TCP transfer.

guaranteeing the minimum MOS, is considered due the mistakes in the measure, as said above.

The packet loss reduction, in percentage, with respect to a reference, is defined in Eq. (2). P_{loss_x} is the percentage of lost packet of the configuration x. $P_{\text{loss}_{\text{Ref}}}$ is the percentage of lost packet of the configuration taken as reference.

$$P_{\text{loss}_{\text{reduction}}} = P_{\text{loss}_x} - P_{\text{loss}_{\text{Ref}}} \quad (2)$$

5.1. No transfer

Figs. 3–5 show, respectively, the MOS values for the whole work session (video-conference), for the video and for the audio (voice) flow if the video bit rate is varied and the QoS scheduling mechanism changed. No disturbance is added in this case. The effect of the RSVP is outstanding: while the quality of the FIFO session drastically deteriorates, the RSVP session always maintains a high quality. In this case, the WFQ scheme too allows getting a good quality because, due to the fair queuing algorithm, low bit rates flows, as voice, which is of main importance for the global evaluation, are

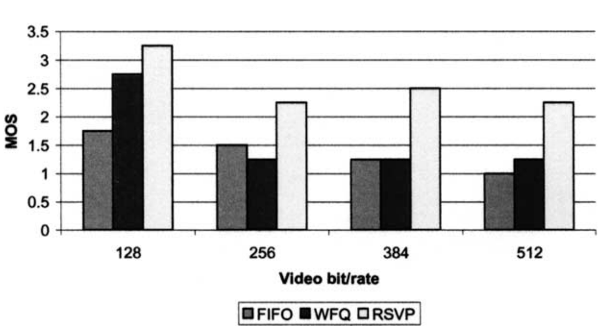

Fig. 9. Video: P-QoS evaluation, TCP transfer.

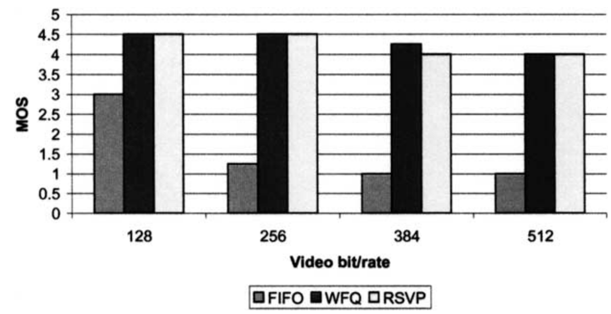

Fig. 10. Voice: P-QoS evaluation, TCP transfer.

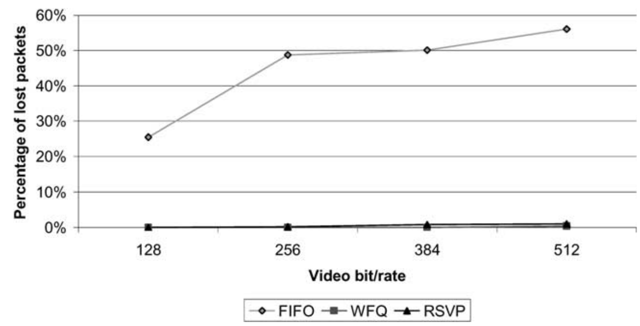

Fig. 11. Voice packet loss, TCP transfer.

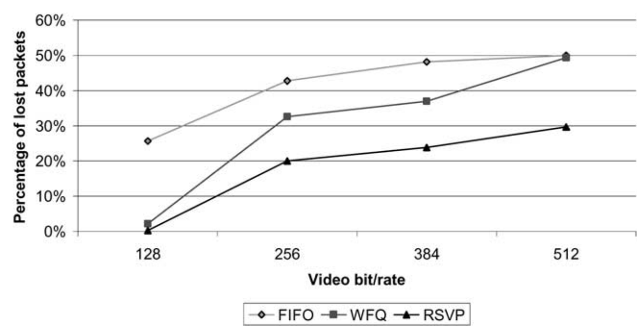

Fig. 12. Video packet loss, TCP transfer.

privileged. The subjective behavior may be justified with objective measures. Fig. 6 reports the percentage of packets dropped for the voice; the configuration is the same as in the previous three figures. The percentage of dropped packets in the FIFO case drastically increases and heavily influences the

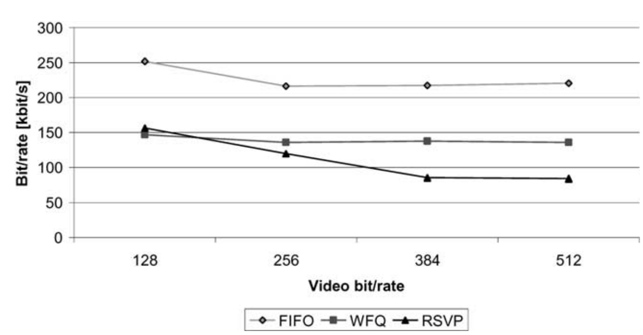

Fig. 13. FTP transfer average bit rate.

Table 7
Gain in percentage, TCP transfer, FIFO as reference

Bit rate	Video-conference		Video		Audio	
	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)
128	33.3	44.4	57.1	85.7	50	50
256	80	140	0	50	260	260
384	125	175	0	100	325	300
512	50	175	25	125	300	300

quality of the comprehension (the overall judgement, in Fig. 3, is very low). The same values concerning video have been shown in Fig. 7. In this case, the number of the lost packets increases also in the WFQ case and only RSVP allows guaranteeing high quality (Fig. 4). Nevertheless, due to the high quality of the audio, the general evaluation (Fig. 3) is not so reduced for WFQ. Table 3 contains the gain in percentage of the WFQ and RSVP configurations, with respect to the simple FIFO, for the tests reported in Figs. 3–5. Table 4 shows the gain of RSVP taking WFQ as reference. Table 5 contains the packet loss reduction of WFQ and RSVP, taken from Figs. 6 and 7; FIFO is the reference. Table 6 reports the same quantity for RSVP, if WFQ is the reference. The performance enhancement, if WFQ or RSVP is used, is outstanding (Table 3), with respect to the FIFO configuration, except for the 128 kbits/s video bit rate case. The large difference in the MOS is motivated by the packet loss reduction (Table 5) guaranteed by WFQ and RSVP. At some extent, this behavior may be expected, even if the quality increase is really huge (up until 350%). The analysis of RSVP taking WFQ as reference (Tables 4 and 6) is more interesting. Even if the RSVP flow protection guarantees a video gain of 62.5% (Table 4, 384 kbits/s), as a consequence of the video packet loss reduction (Table 6), it is not sufficient to influence significantly the overall video-conference evaluation. This is due to the voice performance, which is dominant in this type of service. Only for a higher bit rate (512 kbits/s), the gain, both for video (87.7%) and for audio (12.5%), justifies a meaningful global performance increase (27.3%).

5.2. TCP transfer

It is interesting to investigate the system behavior in presence of an imposed disturbance. Figs. 8–10 contain the same quantities as Figs. 3–5, but an FTP (TCP-based) transfer is performed. The traffic disturbance, identified neither by the FIFO nor by the WFQ algorithm, deteriorates the performance. Only the results obtained with RSVP maintain a good level of quality of service. It is important to note the behavior of WFQ: it allows guaranteeing bandwidth for the voice flow (Fig. 10) but fails to serve the video, which has a so low quality (Fig. 9) to affect the global evaluation (Fig. 8). As in the previous case, the evaluation of the P-QoS may be matched with the objective metrics. Fig. 11 contains the voice packet loss in case of FTP transfer. RSVP and WFQ provide very low losses, while FIFO imposes a high packet dropping even for a limited speed of the video. The percentage of lost packets for the video is reported in Fig. 12. In this case, only RSVP limits the number of packet drops. It is relevant to note that, if the audio flow is always guaranteed by the RSVP algorithm (the bandwidth reserved for video guarantees only one flow, being set to 64 kbits/s, but two voice flows seldom overlap because it would correspond to two speakers in the same time), the video flow is completely guaranteed only for a video bit rate of 128 kbits/s. RSVP really guarantees the flows and makes the effect of the disturbance almost negligible. The disturbance flow is relegated in the residual portion of the bandwidth, which, for the RSVP rules, cannot be reserved at all. That is not true for the other algorithms (WFQ and FIFO), where the TCP-based transfer is treated as the work session flows. The

Table 8
Gain in percentage, TCP transfer, WFQ as reference

Bit rate	Video-conference RSVP (%)	Video RSVP (%)	Audio RSVP (%)
128	25	18.2	0
256	33.3	80	0
384	22.2	100	0
512	83.3	80	0

Table 9
Packet loss reduction, TCP transfer, FIFO as reference

Bit rate	Voice packet loss		Video packet loss	
	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)
128	25.4	25.4	23.5	25.4
256	48.6	48.6	9.9	22.8
384	49.3	49.3	11.2	24.3
512	55.1	55.1	0.7	20.4

Table 10
Packet loss reduction, TCP transfer, WFQ as reference

Bit rate	Voice packet loss RSVP (%)	Video packet loss RSVP (%)
128	0	1.9
256	0	12.6
384	0	13.1
512	0	19.7

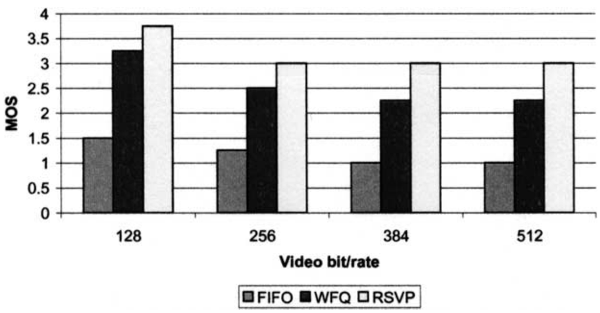

Fig. 14. Video-conference: P-QoS evaluation, UDP transfer.

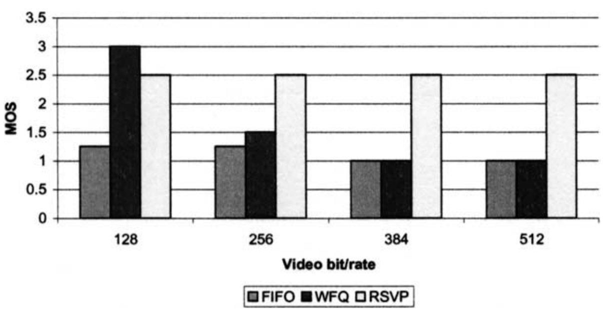

Fig. 15. Video: P-QoS evaluation, UDP transfer.

behavior should be clear from Fig. 13 where the average bit rate of the FTP session is reported for the three schemes considered. Table 7 shows the WFQ and the RSVP gain, taken from the value of the previous three figures (Figs. 8–10), with respect to FIFO. The RSVP gain, with WFQ as reference, is shown in Table 8, while Tables 9 and 10 contain the packet loss reduction (from Figs. 11 and 12), taking, respectively, FIFO and WFQ as reference. In this case, not only the difference (in Table 7) between WFQ and FIFO for video is less remarkable than the previous, non-disturbed case (in Table 3) but also, for high bit rates, the performance of the two mechanisms (WFQ and FIFO) is equivalent. The packet loss reduction is only 0.7%, as reported in Table 9, for 512 kbits/s. Only the use of RSVP allows an outstanding video performance improvement, both having FIFO (Tables 7 and 9) and WFQ (Tables 8 and 10), as reference. So, even if there is no difference between WFQ and RSVP, concerning the audio flow, the high values of video packet loss for WFQ, which are always over 32% in Fig. 12, for high bit rates, heavily affect the global performance. The comparison between Figs. 7 and 12 is particularly meaningful concerning the packet loss impact over the P-QoS and helps map the objective and subjective metrics. Even if the relative 'distance' between WFQ and RSVP does not change significantly (Table 10 versus Table 6), the video packet loss value of WFQ, which is kept relatively low in Fig. 7, arises up much over 30% in Fig. 12. The corresponding P-QoS in Fig. 9 is below 1.5. RSVP helps maintain a packet loss (in Fig. 12) below 30%, which guarantees a MOS value above 2, as clear in Fig. 9 and also by comparing the video packet loss values for WFQ in Fig. 7 with the corresponding MOS results in Fig. 4.

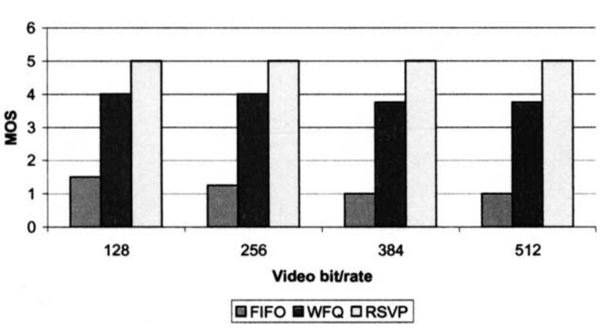

Fig. 16. Voice: P-QoS evaluation, UDP transfer.

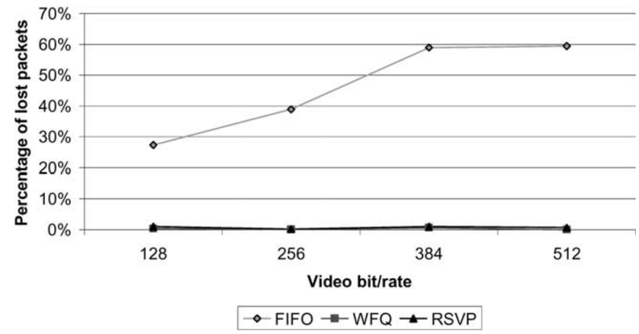

Fig. 17. Voice packet loss, UDP transfer.

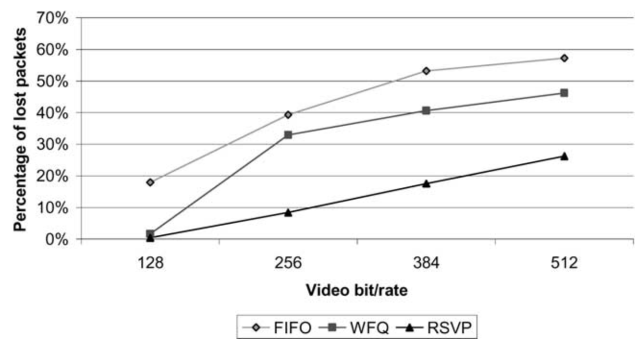

Fig. 18. Video packet loss, UDP transfer.

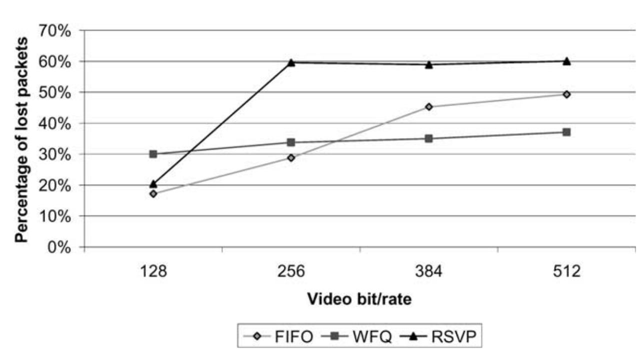

Fig. 19. Disturbance video packet loss.

Table 11
Gain in percentage, UDP transfer, FIFO as reference

Bit rate	Video-conference		Video		Audio	
	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)
128	116.7	150	140	100	166.7	233.3
256	100	140	20	100	220	300
384	125	200	0	150	275	400
512	125	200	0	150	275	400

5.3. UDP transfer

Similar considerations may be done if the disturbance flow is UDP-based, namely a video flow. Figs. 14–18 contain, respectively, the same quantities as Figs. 8–12, but an UDP-based transfer is imposed in this case. The performance is similar to the previous TCP case, but the reduction of quality, for FIFO, is even more evident for low video bit rates (Figs. 14–16, at 128 kbits/s) due to the fact that UDP does not adapt its rate to the network load. As in the TCP case, RSVP reserves bandwidth to the selected flows; the disturbance video flow, which is a non-guaranteed video flow, uses only the residual bandwidth. Its performance is really low, as shown by the RSVP results in Fig. 19, where the percentage of dropped packets is reported for the disturbance video. On the contrary, the results obtained by the other two algorithms (Fig. 19) are not too different from the same quantities measured for the work session video (Fig. 18).

Actually, the similarity of results obtained by using TCP and UDP disturbance flow is due also to the average operation; i.e. averaging the results from two differently connected sites, as Pisa and Genoa, reduces the differences imposed by the different transmission medium. For instance, due to the characteristics of satellite channel [9], the effect of a TCP connection significantly changes. That is not true for UDP. The 'merging' does not allow to appreciate the differences but, in the same time, allow to better identify the general advantages and drawbacks of the mechanisms.

Tables 11–14 contain the same quantities as Tables 7–10, but referred to the UDP-based disturbance case. Comments may be similar as in the previous case (TCP disturbance) but the effectiveness of RSVP (Tables 11 and 12) is worth noting.

Table 12
Gain, UDP transfer, WFQ as reference

Bit rate	Video-conference RSVP (%)	Video RSVP (%)	Audio RSVP (%)
128	15.4	0	25
256	20	66.7	25
384	33.3	150	33.3
512	33.3	150	33.3

Table 13
Packet loss reduction, UDP transfer, FIFO as reference

Bit rate	Voice packet loss		Video packet loss	
	WFQ (%)	RSVP (%)	WFQ (%)	RSVP (%)
128	26.4	26.4	16.3	17.6
256	38.7	38.7	6.38	31
384	58	58	12.6	35.6
512	58.8	58.8	11.1	31

Table 14
Packet loss reduction, UDP transfer, WFQ as reference

Bit rate RSVP (%)	Voice packet loss (RSVP) RSVP (%)	Video packet loss (RSVP) RSVP (%)
128	0	1.3
256	0	24.6
384	0	23
512	0	19.9

6. Conclusions

The paper addresses the problem of guaranteeing a certain level of quality of service and a certain bandwidth reservation for audio and video transmission in a satellite environment. Three different scheduling mechanisms have been tested: a FIFO, a WFQ and a RSVP-based scheme. Real work sessions have been performed to get the results. The quality of service perceived by the users has been measured with a MOS metric; subjective metrics has been associated with objective measures as the packet loss rate and the average transmission rate. For a more detailed investigation the matching of the two results has been reported. It has allowed to completely justify the MOS values obtained and to map the level of user perception with objective metrics of quality. The work sessions have been tested both without any disturbance and with two types of disturbances. A TCP-based traffic, namely FTP session, and an UDP-based traffic, where the audio/video application of interest has been disturbed with a non-guaranteed 256 kbits/s video transmission. The experimental environment adopted, completely private and managed by the authors, was composed of three remote LANs connected through a satellite channel, where Ka-band satellite devices were available and through ISDN, where no satellite device was available. The results allowed getting a full investigation of the system behavior and to tune the various parameters and QoS mechanisms involved so to obtain a good quality perceived by the users and a fully QoS-oriented operative environment by using solutions available in the market. The difference among the different schemes used (FIFO, WFQ and RSVP) is highlighted as well as the measure of the performance improvement due to the application of a bandwidth reservation mechanism.

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