## Abstract

This article describes the activity performed in the three-year project "Integration of Multimedia Services on Heterogeneous Satellite Networks," funded by the Italian Space Agency (ASI) and carried out by the Italian National Consortium for Telecommunications (CNIT). The description of the activity allows highlighting the problems, findings, and solutions in utilizing TCP/IP-based services and applications, such as audio-video transmission and Web browsing, via satellite channels. The article presents an experimental approach to provide QoS guarantees and to improve the performance of TCP over a satellite network based on the TCP/IP suite. The aim is to obtain a proper environment for data, voice, and video transmission oriented to a distance learning service that uses both audio-video applications and Web-based tools. The experimental scenario is made up of three remote LANs. Two of them are connected through a geostationary orbit satellite link in the Ka-band (20–30 GHz). The other LAN, located in a site where no satellite station is available, is connected through an ISDN link at 512 kb/s. Both subjective metrics such as mean opinion score and objective metrics such as throughput and overall transmission time have been used to evaluate the performance of the system and to get proper configurations able to guarantee a high QoS perceived by the users. The Integrated Services approach along with the Resource Reservation protocol were chosen to reserve the network resources at the IP level. A modified version of TCP has been proposed and utilized to reduce the download time during Web browsing and ftp sessions. The measures reported have been obtained by real operative sessions.


# *TCP/IP-Based Multimedia Applications and Services over Satellite Links: Experience from an ASI/CNIT Project*


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The recent evolution of the Internet and the spread of networked multimedia applications have highlighted the need to investigate the techniques, tools, and device configurations to guarantee a certain level of quality of service (QoS) to end users. The characteristics of the network used are extremely important to get to the aim. Some types of networks, such as asynchronous transfer mode (ATM), were designed to support QoS for specific traffic flows. A statistic investigation to verify the availability of the resources to guarantee a fixed level of service is performed before accepting a new call entering the network. A call admission control (CAC) mechanism is used in this case. On the other hand, the Internet is a "best-effort" network (TCP/IP protocols were not designed to provide guaranteed QoS), and it only does its best. It is characterized by heterogeneity from the points of view of both algorithms and management, and physical links. There are high-speed channels, low-speed phone links, wireless multi-access channels, and satellite portions. Moreover, many organizations and providers manage the Internet. In the meanwhile, its development has been incremental, and the number of services and applications are growing more and more.

In this context it is very important to provide guaranteed services over TCP/IP networks. TCP/IP is widely applied, and networks based on it represent a real opportunity to offer user services. See [1] for a recent and complete discussion on the topic.

As a consequence, different approaches have been proposed dependent on the functional level: the network level (IP), transport level (TCP), or application level.

IP is implemented hop by hop, and is responsible for switching and routing data throughout the network. The first step to obtain a QoS guaranteed network based on TCP/IP is to enhance the IP features to offer different services to single traffic flows or groups of traffic flows. Two different approaches have been introduced: differentiated [2] and integrated [3] services. Differentiated services (DiffServ)-based algorithms use the priority fields of the IP packet to differentiate the service offered. Management inside routers is fundamental, and often no resource is allocated in advance. On the other hand, integrated services (IntServ) preallocate resources and reserve a portion of the overall bandwidth for a specific traffic flow. IntServ-based schemes use a signaling protocol, called Resource Reservation Protocol (RSVP) [4], to indicate the bandwidth reservations.

Transport protocols, such as TCP, act on an end-to-end basis; as a consequence, they permit some flow control but cannot guarantee specific requirements to each flow. Nevertheless, their modification and adaptation is precious and may really increase the efficiency of the communication and the quality perceived by the users (as should result from the article).

Approaches acting at the application level are affected by the same drawbacks: they allow setting all the parameters at the application level but delegate the control problem to the network borders (end-to-end). They cannot guarantee any precise requirements, but only adjust the amount of traffic entering the network. They are effective for best-effort traffic but, if the aim is providing service to voice-based applications, they should be used only with great attention. Application-


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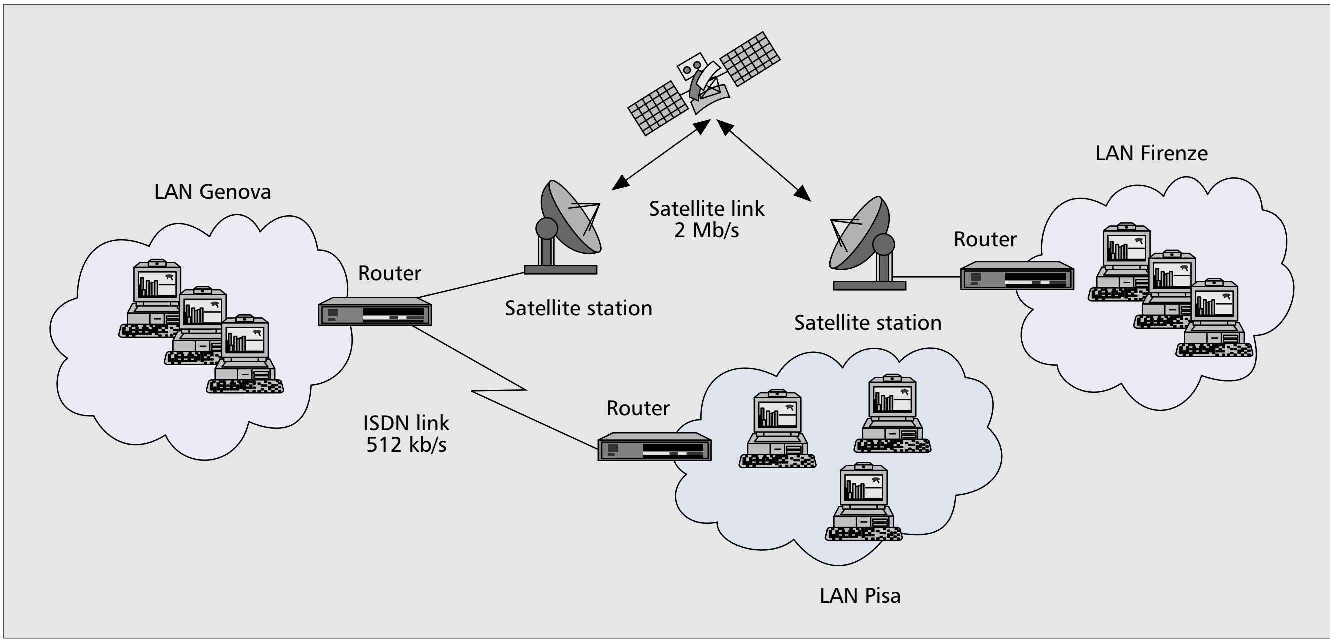



**Figure 1.** *The experimental framework.*

level control schemes are very important because they often take into account the real QoS perceived by the users (perceived QoS, P-QoS), which is frequently ignored when objective metrics are adopted.

The problems envisaged are made worse if a portion of the path is composed by geostationary orbit (GEO) satellite links, whose round-trip delay and general characteristics heavily affect the performance of the protocols at every functional level [5]. The performance of the transport protocol is an issue of particular importance in this environment; if the transport protocol does not offer high performance, the network throughput may become really low and the quality perceived by the users may be poor.

TCP (or UDP, User Datagram Protocol, a simple protocol of the same stack, for audio and video transmission) has been chosen as the transport protocol due to the application environment described above. TCP was not designed for a GEO satellite network environment: the high delay to receive acknowledgments decreases performance and makes the P-QoS really poor. Possible solutions to improve performance of the protocol over satellite links are discussed in this article.

The experimental environment adopted for the present work has the advantage of being completely private and managed by the authors themselves to avoid many problems about the property of different network portions. However, even being a private network, it presents the same problems of a larger TCP/IP network. On the other hand, it allows testing solutions more simply. It is composed of three remote LANs connected through a satellite link (where a Ka-band, 20–30 GHz, satellite channel is available) and through integrated services digital network (ISDN) when no satellite station is available. The IntServ approach has been chosen to reserve bandwidth, and a modified version of the TCP has been introduced to improve performance in file transfers and Web browsing. Mbone tools (session directory, sdr, for multicast session announcement; videoconferencing, VIC, for video; and robust audio tool, RAT, for audio) were used to transmit video and voice. No dynamic flow control is applied at the application level. See [6] for a description of these tools.

Several tests were performed to analyze the effect of bandwidth reservation, scheduling mechanisms, video bit rate, and transport protocol on the performance of the telelearning application. The P-QoS was measured by using a mean opinion score (MOS) method in different situations of traffic load and with different configurations of TCP. All the results were obtained through real work sessions. Both the authors and other people familiar with the tools adopted have contributed to performing the sessions and getting the results.

The article contains a description of the network architecture and the testbed. Issues related to QoS provision in IP-based networks are investigated, and topics of special interest and importance such as IntServ, RSVP, scheduling in routers, and tuning of application parameters are discussed. The main problems related to TCP in a satellite environment are envisaged, and the practical implementations and configurations adopted are presented. Results about audio-video transmission, ftp transfers, and Web browsing are also reported.

## The Network Testbed

Figure 1 shows the experimental scenario. Two LANs, located in Genova and Firenze, are connected through a satellite link at 2 Mb/s; the LAN located in Pisa, where no satellite station is available, is connected to the network in Genova using ISDN. The gateways among the LANs and the external parts (satellite stations or ISDN) are represented by routers, which are network devices operating at the IP level.

The system employs the ITALSAT II (13° EST) satellite in the Ka band (20–30 GHz), which is currently explored for the provision of new services. Each satellite station can be assigned a full-duplex channel with a bit rate ranging from 32 kb/s to 2 Mb/s. The latter was used for the tests. Each satellite station is composed of a satellite modem and a radio frequency device.

The LANs are composed of PCs equipped with a video capture device (H.261 video encoder card). The PCs are the source of the services under test (TCP/IP videoconferencing tools, TCP/IP file transfer and Web browsing).

## IP QoS Issues

Most multimedia services, distance learning in particular, are based on continuous media applications, which are typically sensitive to high end-to-end delay and delay jitter. They depend on the particular real-time network application, and packets that incur more than an $X$ second delay, where $X$ can range from 100 ms to some seconds, are considered useless and dropped by the receiving application, with a relevant decay of P-QoS. These types of applications are typically more loss-tolerant than traditional IP applications: occasional loss, which only causes occasional glitches in the audio and video, can be tolerated by users. On the other hand, traditional IP applications as file transfer and Web browsing may also assume a particular sensitivity to delay transfer (in addition to sensitivity to loss, which is typical for these applications) if applied to distance learning over satellite channels.

In terms of service requirements, multimedia applications, such as real-time audio and video, are very delay-sensitive, and it is necessary to guarantee bandwidth to each real-time traffic flow in order to have high P-QoS. Applications such as file transfer and Web browsing do not require a specific guaranteed bandwidth, but their performance may be drastically improved by protocol modifications for special environments, such as modified TCP introduced over satellite links.

Concerning audio and video UDP-based transmission, the integrated services (IntServ) approach, which was developed by the Internet Engineering Task Force (IETF) IntServ Working Group to provide specific QoS guarantees to individual application sessions, was implemented within the testbed to optimize network resource utilization. The IntServ network architecture includes a real-time service that provides functions to reserve bandwidth for video and audio flows within an IP network. Because of routing delays and congestion losses, real-time applications do not work well on a best-effort IP network where packets belonging to various network flows are multiplexed together and queued for transmission at the output buffers associated with a link, according to a FIFO scheduling discipline.

The IntServ architecture and the use of RSVP within this framework have received much attention in the literature for a long time. Reference [7] defines an extension of the TCP/IP architecture and protocols to support both QoS guaranteed and nonguaranteed services. An example of design and implementation of a protocol architecture based on IntServ is contained in [8], which also lists many references related to the topic in a dedicated section.

To support the IntServ model, an IP router must be able to provide an appropriate QoS to each flow. The router function that provides different QoS is called the *traffic control module* and consists of the following components.

***Packet Scheduler*** — This software module manages the forwarding of different packet streams in hosts and routers, based on their service class. It uses queue management techniques and various scheduling algorithms. The packet scheduler must ensure that the packet delivery service is carried out to meet the QoS parameters requested by each flow. A scheduler can also police or shape the traffic to conform to a certain level of service.

***Packet Classifier*** — This module identifies the packets belonging to an IP flow that will receive a certain type of service in hosts and routers. Each incoming packet is mapped by the classifier into a specific class, based on the source and destination IP address, and the source and destination TCP/UDP port contained in the header of the IP packet.

***Admission Control*** — The admission control module implements the decision algorithm a router uses to determine if there are enough routing and network resources to accept a reservation request coming from a new flow without damaging the service level guaranteed to the flows already accepted.

If the QoS request of a new flow is accepted, the reservation instance in the router assigns the necessary resources to guarantee the requested QoS to the flow.

Thus, two key features lie at the heart of an IntServ architecture:

- Each router is required to know the amount of resources (buffer, link bandwidth) already reserved for ongoing sessions.
- A session requiring QoS guarantees must be able to reserve enough resources at each network router along the source-destination path to ensure that its QoS requirements are met.

In order to determine whether the available resources are sufficient to meet the QoS requirements, a session must declare its QoS requirements and characterize the traffic it will send into the network. Even if admission control is part of the IntServ model and is reported for the sake of completeness, no admission control has been performed in this work.

Best-effort routers use first in first out (FIFO) queuing with the result that the traffic is transmitted in the order received without regard for bandwidth consumption or associated delay. It is necessary to use a different scheduling scheme to provide guaranteed service. Weighted Fair Queuing (WFQ) is an automatic scheduling discipline providing fair bandwidth allocation to all network traffic. With standard WFQ, packets are classified per flow. Packets with the same source IP address, destination IP address, source TCP/UDP port, and destination TCP/UDP port belong to the same flow. WFQ allocates an equal share of the bandwidth to each flow. Flow-based WFQ is also called Fair Queuing (FQ) because all flows are equally weighted.

RSVP is the signaling protocol designed by the IETF IntServ Working Group to allow applications to reserve network bandwidth dynamically. 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 specific data flow. If RSVP is used in conjunction with WFQ to set up the packet classifier and packet scheduler parameters, it is possible to provide differentiated and guaranteed QoS services (i.e., to fix the link capacity assigned to specific traffic flows).

Within the testbed, taking advantage of some router features, video and audio applications have not been modified to support RSVP signaling protocol: the routers directly connected to the LAN of the transmitting hosts were configured as if they were receiving signaling messages from them.

## *TCP/IP in a Satellite Environment*

Use of the TCP/IP stack in satellite environments is really important: many applications are based on this protocol family, and any connection with the Internet of the future will imply the use of TCP/IP. The network used heavily affects the behavior of the transport protocol; even in a satellite environment, the problems are different if a low earth orbit (LEO), medium earth orbit (MEO), or GEO satellite system is used. Issues related to each environment are listed in [5]. A geostationary system, where "there is an inherent delay in the delivery of a message over a satellite link due to the finite speed of light and the altitude of communications satellites" [5], is approached in this work. The round-trip time (RTT) is above 500 ms and the bandwidth-delay product is large. The product measures the maximum number of bits the network can hold and, concerning TCP, the maximum amount of data that can be sent before receiving an acknowledgment. The bandwidth-delay product influences TCP performance: the arrival of data and acknowledgments is slow, and the overall throughput of the connection is thus low. On the other hand, there are also positive aspects: the

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

■ **Table 1.** *Application parameters.*

RTT is approximately constant, connectivity is guaranteed in geostationary systems (not in LEO, for instance), and the transmission errors measured are very low, at least in the testbed used. Moreover, if the application field has a limited extension (i.e., a limited number of satellite links or a small number of hops), as in the testbed, the characteristics of the links and of the devices traversed are well known and the congestion aspects may be treated differently than in a large cabled network. So, even if the performance of TCP is poor, improvements can be obtained by properly tuning some parameters and modifying algorithms.

The problem of improving TCP over satellite has been investigated in the literature for some years: see [9] for a first overview on the topic. More recently, [10] provides a summary of improved TCP versions as well as issues and challenges in satellite TCP. Reference [11] lists the main limitations of TCP over satellite and proposes many possible solutions. A recent tutorial on the topic is contained in [12]. Reference [13] shows an extensive analysis of the TCP behavior by varying parameters as buffer size and initial congestion window. Many works (e.g., [9, 14]) propose architectures to provide Internet services via satellite, and introduce relay entities to split the TCP connection and isolate specific parts of the network that deserve particular attention concerning the transport level (e.g., satellite links).

Many national and international projects (listed extensively in [13]) in Europe, Japan, and the United States involve satellite networks and applications. In particular, some of them, or part of others, are aimed at improving performance at the transport level.

There are some ways to improve TCP over satellite; a possibility is acting on the protocol parameters and algorithms to tune and modify them to mitigate the negative effect of the satellite channel and improve performance. That is the choice made in this article.

The short summary in the following is intended to focus on the TCP characteristics of concern in this article. The parameters are substantially set by following the standard and the notation introduced in [15].

The slow start phase begins setting:

- The congestion window (cwnd) to 1 segment ($1 \cdot$ smss, where smss, measured in bytes, stands for sender maximum segment size)
- The slow start threshold (ssthresh) to a very high value (infinite)

A parameterization of the initial congestion window (IW) is followed by setting cwnd = IW $\cdot$ smss. The buffer size (buf) at the transmitter and receiver, set to 64 kbytes in the TCP currently used, has been parameterized too.

At each received acknowledgment (ACK), cwnd is increased by $1 \cdot$ smss (i.e., "ACK → cwnd = cwnd + $1 \cdot$smss").

Other important TCP algorithms such as congestion avoidance, fast retransmit, and fast recovery are untouched. The selective ACK (SACK) mechanism is utilized.

In a geostationary environment, cwnd needs more time to grow than in cabled networks. If, for example, just one segment was sent, it takes at least one RTT to be acknowledged. The throughput is very low, even in the slow start phase. This heavily affects the performance of applications based on TCP. The article reports an analysis and a tuning of the receiver/transmitter buffer space (buf) and the initial congestion window (IW).

## Implementation and Configuration

The scientific aspects highlighted in the previous sections were used as guidelines to configure and implement the testbed.

Due to the high costs of a satellite broadband network, a real but downscaled environment, where a few stations play all the roles found in a larger satellite network designed for distance learning applications, has been built. The main steps necessary to prepare the testbed for the experiments have been the following:

- Identification of the service and application requirements
- Configuration of the devices and protocols

### Identification of the Service Requirements

The following service requirements have been identified.

***Multicast IP Support*** — Since distance learning is a typical one-to-many or many-to many application, complete IP multicast support is necessary to better exploit satellite capabilities and network resources, mainly concerning bandwidth utilization.

Multicast is the act of sending a message to multiple receivers using a single local transmit operation. In a wide area network, each host that wishes to join a multicast session must first inform the local multicast router by using the Internet Group Management Protocol (IGMP); then the local router can interact with other routers to receive multicast packets. A multicast routing protocol, such as Protocol Independent Multicast (PIM) or Distance Vector Multicast Routing Protocol (DVMRP), is necessary. See [6] for a detailed description of multicast protocols.

IGMPv2 was activated at the host level, and PIM dense mode (PIM-DM) was configured within each router in the testbed.

***QoS Support*** — The router is the device located at the edges of each satellite or ISDN link, as shown in Fig. 1. In order to control the utilization of network resources, it is necessary to activate a scheduling technique associated with the routing function.

In particular, three different scenarios, respectively corresponding to FIFO scheduling, FQ scheduling, and WFQ scheduling with RSVP, have been identified to test and evaluate the performance of the applications.

***Application Requirements*** — Distance learning is a multimedia application basically composed of two services: multicast videoconferencing and multicast data dissemination. MBONE tools (VIC, RAT, SDR) were used concerning the former. The settings of the application parameters used in the experiments are summarized in Table 1 for each tool.

### Configuration of Devices and Protocols

To guarantee the QoS requirements requested by audio and video flows, it is necessary to carefully plan the configuration of FQ and RSVP within each router.

- **FQ configuration**

When FQ is configured for an interface, three parameters can be set:

–*Congestive-discard threshold*: Represents the number of packets allowed in each queue. When this threshold is reached, new packets arriving are discarded.

–*Dynamic queues*: The number of dynamic queues used for best-effort conversations.

–*Reservable queues*: The number of queues that can be reserved.

- **RSVP configuration for multicast video and audio sessions**

RSVP was enabled on each router interface, and the maximum bandwidth that could be reserved for each flow was specified. As far as multicast video and audio sessions are concerned, each router connected to a LAN with multicast session members has been configured in order to meet the reservation requests received.

Two different RSVP reservations have been set on each network router:

–Concerning the audio session, since just one sender should transmit data at any given time, a single reservation, which can be applied to any sender belonging to the group identified by the multicast address of the audio session, has been used. In more detail, a guaranteed bandwidth reservation of 64 kb/s has been set up on each router.

–In order to provide proper bandwidth reservation for a video session, it is necessary to take into account that, if RSVP is used, the default maximum bandwidth that can be reserved on each router interface is 75 percent of its available bandwidth, and in a videoconferencing system with *N* stations, each station may simultaneously receive *N* – 1 video flows. Thus, a maximum guaranteed bandwidth reservation of 320 kb/s, which can be applied to any flow characterized by the destination port number and IP address of the multicast video session, was installed. The value 320 kb/s is 75 percent of the value (512 – 64) kb/s. 512 kb/s is the bandwidth of the ISDN link and 64 kb/s that assigned to the audio flow.

- **TCP configuration**

The experimental activity carried out within the ASI-CNIT project, aimed at the evaluation and enhancement of TCP performance over a satellite link [13], led to the selection of some meaningful configurations to be adopted for TCP-based applications.

The results reported in the next section have been obtained by choosing the TCP IW and buffer size (buf) values listed in Table 2.

| Initial congestion window (IW) | Buffer (buf) |
|---|---|
| 1 | 64 |
| 6 | 64 |
| 6 | 320 |
| 1 | 320 |
| 2 | 320 |

**Table 2.** *Initial congestion window and buffer size.*

The first three configurations were also used in the measure of P-QoS during the Web browsing of the learning sessions.

## Results

The results have been structured into two separate parts. The first is dedicated to the performance evaluation of UDP-based applications (i.e., audio and video transmission). The second focuses on such TCP-based applications as file transfer and Web browsing. The two satellite telelearning system components, audio-video and TCP-based sessions, were distinguished because, except for the teaching/learning aim, the two environments have little in common. The transport protocol is different, as well as the QoS requirements. A guaranteed bandwidth is topical to offer an efficient audio-video service, and intervention at the IP level is necessary. File transfer and Web browsing may be strongly improved by modifying only the transport protocol (TCP, in this case). This approach reflects on the presentation of the results: as far as UDP-based applications are concerned, the focus is mainly on the measure of P-QoS. Concerning TCP applications, the modifications of the TCP in order to improve two objective metrics (overall transfer time and throughput) are presented. The effect of the performance improvement on users is then measured by applying modified versions of TCP to a Web learning session implying file transfer and Web browsing.

### *UDP-Based Audio-Video Transmission*

The results presented are aimed at analyzing the performance of the overall telelearning videoconference system. The whole system was evaluated by varying both the video bit rate and the scheduling strategy. Audio bit/rate is fixed at 64 kb/s. Four different values (128, 256, 384, 512 kb/s) were used for the video. The three configurations presented in the previous section were chosen for queue management in the routers: the no fair queuing FIFO, FQ, and RSVP-based schemes (WFQ with RSVP). The work sessions were tested both without any disturbance and with two types of background traffic: TCP-based traffic, namely, FTP sessions, where the reference version (IW = 1 – buf = 64 kbytes) of TCP was used, and UDP-based traffic, where the audio/video application of interest is jammed with a nonguaranteed 256 kb/s video transmission. The aim is to investigate system behavior and tuning the various parameters to obtain good P-QoS.

Many measures have been performed by utilizing a mean opinion score (MOS) method to evaluate real user perception. The score ranges from 1 (bad) to 5 (very good). All the measures were averaged to get an evaluation of the overall system. On the other hand, this approach has some drawbacks: the single measures lose their individual peculiarity; for instance, there is no distinction among different sources or link types (i.e., satellite or ISDN). Nevertheless, too many results would not have obtained the effect of a better clarity. One of the aims of the article is evaluation of the overall work sessions; the results reported should get to the point.

Figure 2 shows the MOS values for the whole work session (videoconference) if the video bit rate is varied and the scheduling mechanism changed. No background traffic is added in this case. The effect of the RSVP guarantees is outstanding: while the quality of the no fair queuing session drastically

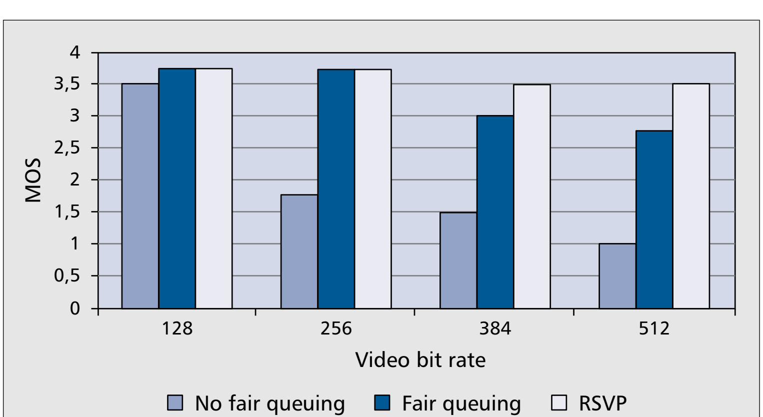



**Figure 2.** *Videoconference: P-QoS evaluation, no background traffic.*

deteriorates, the RSVP session always maintains high quality. In this case, the FQ scheme also allows good quality because, due to the FQ algorithm, low-bit-rate flows (i.e., voice), of major importance for global evaluation, are privileged.

It is interesting to investigate the system behavior when background traffic is added. Figure 3 contains the same quantities as Fig. 2 when a TCP-based FTP transfer is performed. The jamming traffic, which is individuated by neither the no fair queuing nor the FQ algorithm, deteriorates the performance. Only the results obtained with RSVP maintain a certain level of QoS. FQ allows guaranteeing bandwidth for the voice flow but fails to serve the video, which provides low enough quality to affect the global evaluation (Fig. 3).

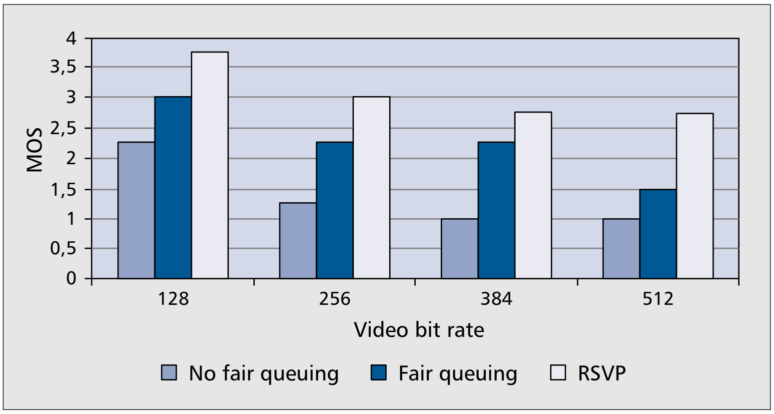



**Figure 3.** *Videoconference: P-QoS evaluation, TCP transfer.*

Similar considerations may be made if the background traffic is UDP-based, such as a video flow. Figure 4 contains the same quantities as Fig. 3, but a UDP-based transfer is completed in this case. The performance is similar to the previous TCP case, but the reduction of quality for the no fair queuing scheduling discipline is even more evident due to the fact that UDP does not adapt its rate to the network load. As in the TCP case, RSVP reserves bandwidth for the important flows; the disturbance video flow, which may also be considered a nonguaranteed video flow over the same network, uses only the residual bandwidth.

## *TCP-Based Transmission: File Transfer and Web Browsing*

This subsection is divided into two parts. The first is dedicated to summarizing the most meaningful results about the behavior of TCP by varying the value of IW (measured in bytes; the notation IW = 1 means IW = 1· smss) and of the buffer dimension, intended as the memory availability in bytes for source and destination, which is kept equal. It is identified with the variable buf in the following. The application used to get the results is a simple ftp-like one, which allows transferring files of variable length (*H* bytes in the following) between the two remote sites. A file of relevant dimension (*H* = 2.8 Mbytes) and a small file (*H* = 100 kbytes) were used to perform the tests. The choice between them is indicated in the text. The presence of more than one connection in the network is obtained by consecutively activating a fixed number (indicated as *N* in the following) of connections (transfers of files of 100 kbytes, in this case). Due to the high speed of the activation, the effect is loading the testbed with *N* connections at the same time, even for transfers of short files.

The study has considered two different application scenarios: single connection and multiple connections. The first case implies that only one connection at a time is in the network; from a practical viewpoint it may represent access to a database of a small private network where just one customer, or very few customers, access the network at the same time. Another example may be a file of commands required by a remote control system. Speed is essential in both cases.

The second case is very common: it represents the situation where many users at a time access the information available on a Web server.

***TCP Modifications*** — The TCP commonly used in cable networks sets IW to 1 and buf to 64 kbytes. The reference configuration is (buf = 64 kbytes, IW = 1). In this situation the increase in speed is very slow, the TCP is drastically blocked by the satellite delay, and the transmission window cannot increase its length because the buffer dimension represents a bottleneck for the system. The buffer size is very important for the performance of the system; it rules the congestion window by imposing a restriction to its increment. A short buffer drastically limits performance, but an excessively long buffer makes the system congested. When the system is congested, the throughput is strongly reduced even if the efficiency is high at the beginning of the connection. A correct choice of the buffer size and of the congestion window is very important to get better performance. Table 3 allows summarizing the effect of the parameters tuning performed. The table contains the combination of the two parameters analyzed (IW and buf), the time required for the overall transmission and the gain in percentage obtained with respect to the reference configuration (IW = 1, buf = 64 kbytes) with *H* = 2.8 Mbytes. The gain in the overall transmission time (up to 71.63  percent) is mainly due to the buffer size, which may represent a real bottleneck for the system. The gain is computed as follows: if is the reference transmission time ($T_{REF}$ = 49.21 s, in Table 3) and *T* is a generic transmission time, the percentage gain is defined as

$$\%\mathrm{Gain} = \begin{cases} \dfrac{T_{REF} - T}{T_{REF}} \cdot 100, \text{ if } T < T_{REF}. \\ 0, \end{cases}$$

If $T_{REF} \le T$, there is no gain. For example, referring to Table 3, if $T$ = 13.96 s, % Gain = (49.21 – 13.96)/ 49.21 = 71.63 %.

Figure 5 shows the average throughput per connection and compares the behavior of different TCP configurations vs. the number of active connections *N* for a 100 kbyte transfer. The

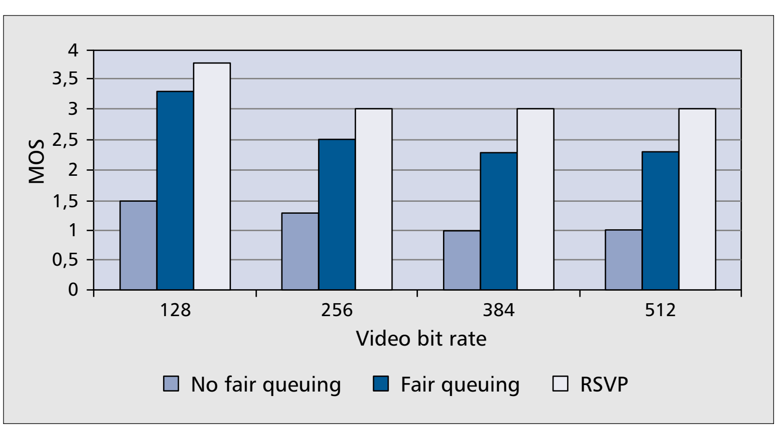
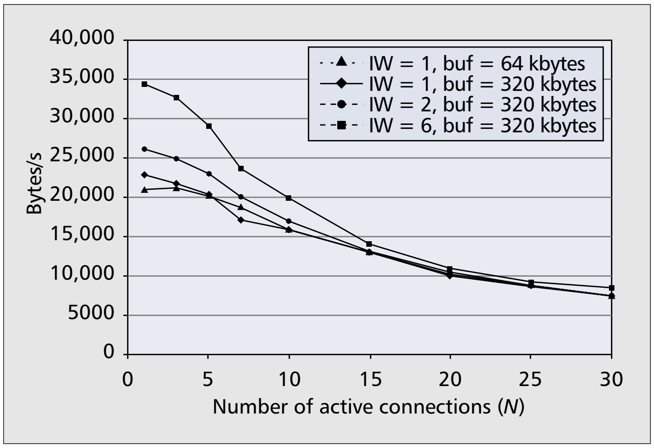



**Figure 4.** *Videoconferencing: P-QoS evaluation, UDP transfer.*

| IW , buf (kbytes) | Transmission time (s) | Throughput (bytes/s) | Gain (%) |
|---|---|---|---|
| 1, 64 | 49.21 s | 56,665 | – |
| 6, 64 | 47.55 s | 58,889 | 3.37 % |
| 1, 320 | 15.87 s | 175,691 | 67.75 % |
| 6, 320 | 13.96 s | 199,812 | 71.63 % |

■ **Table 3.** *A comparison of TCP configurations by varying the initial congestion window and buffer size,* H = *2.8 Mbytes.*




■ **Figure 5.** *Throughput (bytes/s) versus the number of active connections (N), H=100 kbytes.*

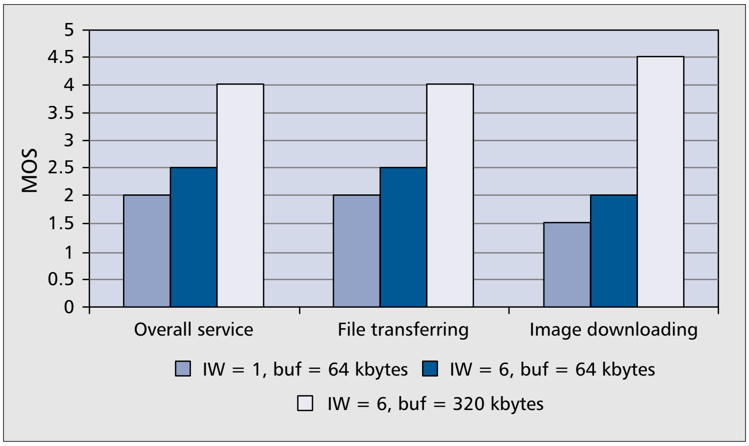
test is aimed at verifying the effect of tuning in the presence of a network loaded with multiple connections. Four configurations are taken into account: the reference configuration (IW = 1, buf = 64 kbytes); (IW = 1, buf = 320 kbytes), where only the buffer is varied; and two configurations where both buf and IW are increased: (IW = 2, buf = 320 kbytes) and (IW = 6, buf = 320 kbytes).

It is important to note that the higher traffic load does not cancel the performance improvement, due to modified values of IW and buf. The gain in throughput is evident up to 10 active connections; for values larger than 10, the traffic load due to the number of connections in progress makes TCP insensitive to the modifications introduced.

***TCP-Based Applications: P-QoS Measure*** — This subsection reports the results of a Web telelearning session. The Website is located by the LAN in Firenze (Fig. 1), and is composed of files and figures dedicated to a specific topic. The students remotely access the Website from Genova and Pisa, where each student has a dedicated computer to perform the operation. The reported measures, obtained with MOS methodology, represent the average of users' opinions. The maximum number of computers allowed for each remote LAN was limited to five. After this limit, as is clear from the results reported previously, there is no clear advantage in using modified TCP. Even if the effect of the modification is very different if the user is located in Genova, where a full 2 Mb/s channel is available, or in Pisa, where the maximum transfer capability is 512 kb/s, the measures have been averaged. The choice is aimed at evaluating the overall Web telelearning system and simplifying interpretation of the results.

The tests have been performed using three configurations: (IW = 1 – buf = 64 kbytes), (IW = 6 – buf = 64 kbytes), and (IW = 6 – buf = 320 kbytes). The results refer to the evaluation of both the overall Web telelearning service and the two main components of the service: file transfer and image downloading. Even if the two components cannot be distinguished from the network point of view because they both represent a data transfer as used in the previous subsection, the effect on user perception might be different.

Figure 6 contains the MOS measures concerning the overall service, file transferring, and image downloading by varying the TCP configuration of the computers involved. The improvement in P-QoS with the (IW = 6 – buf = 320 kbytes) modified TCP version is outstanding.




■ **Figure 6.** *P-QoS evaluation: Web telelearning.*

## Conclusions

This article summarizes the experience of three years in the ASI/CNIT project Integration of Multimedia Services on Heterogeneous Satellite Networks. One of the main activities of the project as well as an important application where new satellite technologies in the Ka-band may be applied is tele-education. Remote teaching/learning is essentially based on two components: audio-video UDP-based transmission and TCP-based telelearning Web browsing. The need to guarantee a good level of quality of service to users of the system for both components has implied the knowledge and investigation of the last findings and of the current research about TCP/IP over satellite, which are focused on and highlighted in the article. Concerning audio and video transmission, the problem of guaranteeing a certain level of QoS and a certain bandwidth reservation has been approached. In more detail, three ways of supporting QoS in an IP satellite environment aimed at videoconferencing and remote distance learning are tested in this article: FIFO, Fair Queuing, and RSVP-based. Real work sessions were performed to get the results. The QoS perceived by users was measured with a MOS metric. The work sessions were tested both without any background traffic and with two types of background traffic: TCP-based traffic, generated by ftp sessions using a non-modified TCP version, and UDP-based traffic, a nonguaranteed 256 kb/s video transmission that jammed the audio/video application of interest.

The need to improve Web telelearning service has compelled investigation of the performance of TCP over GEO satellite links, where round-trip time is above 500 ms.

TCP was not designed for these network characteristics: the high delay to receive acknowledgments decreases the performance and makes the quality perceived by users really poor. The article includes an analysis of the TCP behavior by varying the buffer size and congestion window of the protocol to better adapt to large-delay satellite links. The improvement obtained is outstanding. A gain above 70 percent is reached with respect to the TCP reference configuration when only one connection at a time is routed in the network. The analysis shows that the solutions proposed also maintain a satisfying gain in a multiconnection environment. The new algorithm, tested in a real satellite testbed, was implemented by modifying the operating system and was operatively used in Web telelearning sessions during which the QoS perceived by users was measured.

During both UDP and TCP-based applications, the experimental environment, completely private and managed by the authors themselves, was composed of three remote LANs connected through a satellite channel when Ka-band satellite stations were available, and through ISDN when no satellite station was available.

The results allowed a full investigation of the system behavior, and tuning of the various parameters and QoS mechanisms involved to obtain good quality perceived by users.

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