A Quality of Service Guarantee in IP Satellite Environment: experimental experience in the CNIT-ASI Project "Integration of Multimedia Services on Heterogeneous Satellite Networks"

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Abstract - **The paper presents an experimental approach to provide a guaranteed Quality of Service (QoS) over a satellite network based on the Internet Protocol (IP). The results obtained represent part of the experimental activity carried out during the second year of the Project "Integration of Multimedia Services on Heterogeneous Satellite Networks", called "ASI-CNIT Project". Both subjective metrics as Mean Opinion Score (MOS) and objective metrics, as video and voice packet loss, jitter and transmission rate have been used to investigate the topic and to get proper configurations able to guarantee an high quality of service perceived by the users (PQoS – Perceived Quality of Service). The Integrated Services approach along with the ReSerVation Protocol (RSVP) has been chosen to reserve the network resources. The measures reported have been obtained by real operative sessions.**

I. Introduction

The recent evolution of the Internet and the widespread of networked multimedia application have highlighted the necessity of investigating the techniques, the tools and the device configuration to guarantee a certain level of Quality of Service (QoS) to the end users.

Different approaches have been proposed in dependence of the functional level on which to act: the network level (IP), the transport level (TCP) or the application level.

Concerning the IP, two different point of views have been introduced: Differentiated Services [1], based on priority fields to differentiate the service offered and Integrated Services [2] which relies on a signalling protocol called RSVP [3] to notify the bandwidth reservations.

The problems envisaged are made worse if a portion of the path is composed by Geostationary Orbit (GSO) satellite links, whose round-trip delay and general characteristics heavily affect the performance of the protocols at every functional level [4].

The experimental environment is composed of three remote LANs connected through a satellite channel, where a Ka-band satellite device is available and through ISDN when no satellite device is available. The Integrated Service (IntServ) approach has been chosen to reserve bandwidth. Mbone tools (sdr [5] for multicast session announcement, VIC [6] for video and RAT [7] for audio) have been used to transmit video and voice. The application envisaged is distance-learning.

The paper is structured as follows. The next section contains the description of the network testbed. Section III highlights the issues related to the QoS provision in IP based networks. The practical implementations and configurations adopted are presented in section IV. Section V contains the results and section VI the conclusions.

II. Network Testbed

The experimental scenario where the tests have been performed is reported in Fig. 1: two remote LANs, located in Genova and Prato, are connected through a satellite link at 2 Mbit/s; the LAN located in Pisa, where no satellite device is available, is connected to the LAN located in Prato by using ISDN.

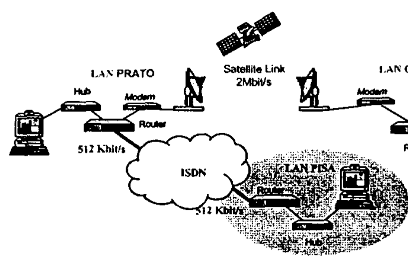

Fig. 1. Experimental framework

The system employs the ITALSAT II (13° EST) satellite, providing a country-wise coverage in the single spot-beam on Ka band (20-30 GHz). The overall bandwidth is 36 MHz. Each satellite station can be assigned a full-duplex dedicated traffic channel with a bit-rate ranging from 32 kbit/s to 2 Mbit/s. Satellite sites are equipped with the following components: satellite modem, radio-frequency device, IP router and application PCs. These latter are the source of the services under test (TCP/IP Video-conferencing tools,

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TCP/IP file transfer and terminal applications, Multicast applications and remote access to scientific instruments). Other sites are equipped with an IP router and a number of Application PCs.

III. IP QoS ISSUES

The Integrated Services (IS) approach, which was developed by the IntServ IETF Working Group [3] to provide specific quality of service guarantees to individual traffic flows, has been implemented within the CNIT-ASI intranet to optimize network resources utilization. To support the Integrated Services model, an IP router must be able to provide an appropriate QoS to each flow. Two key features lie at the heart of an IntServ architecture:

a) each router is required to know the amount of resources (buffer, link bandwidth) already reserved for on-going sessions;

b) a session requiring QoS guarantees must first be able to reserve enough resources at each network router along the source-destination path to ensure that its QoS requirements are met.

Best-effort router use FIFO queueing with the result that the traffic is transmitted in the order received without regard for bandwidth consumption or associated delay. Therefore, to provide a guaranteed service it is necessary to use a different scheduling scheme. WFQ (Weighted Fair Queueing) is an automatic scheduling discipline providing fair bandwidth allocation to all network traffic. WFQ applies priority, or weights, to identified traffic streams according to the bandwidth each conversation is allowed relative to other conversations. RSVP is the signaling protocol designed by the IETF IntServ Working Group for allowing applications to dynamically reserve network bandwidth. 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 Weighted Fair Queueing to set up the packet classifier and the packet scheduler parameters needed to the reserved flows, it is possible to provide differentiated and guaranteed QoS services, that is to fix the link capacity to be assigned to specific traffic flows.

IV. IMPLEMENTATION AND CONFIGURATION

A. *Identification of the target service requirements*

The following target service requirements have been identified:

Multicast IP support

As distance learning is a typical one-to-many or many-to many application, a complete IP multicast support is necessary to better exploit network resources, mainly as it concerns bandwidth utilization. In a wide-area network, each host, wishing to participate to a multicast session, must first inform its local multicast router of its desire to join a group using the Internet Group Management Protocol (IGMP) [8]; then the local router can interact with other routers to receive multicast packets using a Multicast routing protocol. Therefore, IGMPv2 has been activated at host level and PIM [8] Dense Mode has been enabled and configured within each network router.

QoS Support

As it concerns the QoS support, this has been accomplished with a series of protocols/techniques specific to each link type between the nodes of the network. A router is the device located at the edges of each satellite or ISDN link. In order to control network resources utilization, it is necessary to activate a scheduling technique associated with the routing function. In particular, the following operating conditions for testing the QoS support have been identified:

FIFO Scheduling

WFQ Scheduling

WFQ Scheduling with RSVP

c) Application requirements

The last element that influences the test-bed setup is the application requirement definition.

Distance Learning is a multimedia application basically composed of two services: multicast video-conferencing and multicast data dissemination. As it concerns the former, MBONE tools (VIC, RAT, SDR), have been used. The settings of the application tools parameters used in the experiments are summarized in Table I.

TABLE I
APPLICATION PARAMETERS

Tool	Parameter	Value
VIC	Motion Compensation Quality	10
	Video Encoding	H.261
	Frame Rate	15 fps
	Image Format	CIF (352x288)
	Bit-rate	128,256,384,512 kbit/s
RAT	Audio Encoding	PCM - 64kbit/s
	Resolution	16 bit
	Audio Bandwidth	8 kHz

B. *Configuration of devices*

To guarantee the quality of service requirements requested by audio and video flows, it is necessary to carefully planning the configuration of Resource Reservation Protocol within each router. RSVP has been enabled on each router interface and the maximum bandwidth that can be reserved to each flow has been specified. Two different RSVP shared reservations have been set on each network router:

concerning the audio session, as a very limited number of senders are transmitting data at any given time, it has been used a single reservation that can be applied to any sender belonging to the group identified by the multicast address of the audio session. In particular, a guaranteed bandwidth reservation with a rate of 64 kbit/s have been set up on each router;

in order to provide a proper bandwidth reservation for the video session, it is necessary to take into account that the default maximum bandwidth reservable on each router interface is up to 75% of its bandwidth available and that in a videoconferencing system with N stations, each station may simultaneously receive N-1 video flows. Therefore, it has been installed a maximum guaranteed bandwidth reservation with a rate of 320 kbit/s that can be applied to any flow characterized by the destination port number and IP address of the multicast video session.

V. RESULTS

The results presented aim at evaluating the performance of the overall tele-working, distance learning system in each of its component. The whole system is evaluated along with audio and video performance by varying both the video bit/rate and the QoS guaranteeing strategy. Audio bit rate is fixed at 64 kbit/s (PCM). The video bit-rate assumes four different values: 128, 256, 384, 512 kbit/s. The three configurations presented in section IV have been chosen for the queue management in the routers: the no fair queueing, FIFO scheme, the fair queueing scheme, where a fair division of the bandwidth among all the traffic involved is guaranteed; an RSVP - based scheme, where the voice is always guaranteed with 64 kbit/s and the video is guaranteed if possible, depending on the residual bandwidth; namely, a maximum of 320 kbit/s, which represents the maximum possible bandwidth to be allocated for video in the configuration tested, has been guaranteed for video traffic. The work sessions have been tested both without any disturbance and with two types of jamming: a TCP - based traffic, namely, ftp sessions and an UDP - based traffic, where the audio/video application of interest is jammed with a non-guaranteed 256 kbit/s video transmission. The aim is investigating the system behavior and tuning the various parameter to obtain a good quality perceived by the users.

Many measures have been performed, both utilising a MOS method to evaluate the real user perception and adopting objective metrics as the packet loss rate, the transmission bit rate and the jitter (not presented in this work). To simplify the evaluation of the results only the measures obtained in the two remote sites (Genova and Pisa) have been considered. All the measures have been averaged so to get an evaluation of the overall system. Fig. 2, Fig. 3 and Fig. 4 show, respectively, the MOS values for the whole work session (videoconference), for the video and for the audio (voice) flow if the video bit/rate is varied and the QoS mechanism changed. No disturbance is imposed in this case (no transfer). The effect of the RSVP guarantees is outstanding: while the quality of the "no fair queueing" session drastically deteriorates, even for not so high video bit/rates, the "RSVP" session always maintains a high quality. In this case, the fair queueing scheme too allow to get a good quality because, due to the fair queueing algorithm, low bit/rates flows, i.e. voice, which is of main importance for the global evaluation, are privileged. The subjective behaviour may be justified with objective measures. Fig. 5 reports the percentage of lost packets for voice; the configuration is the same as the previous three figures.

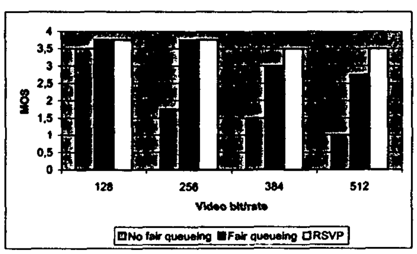

Fig. 2. Videoconference: PQoS evaluation, no transfer.

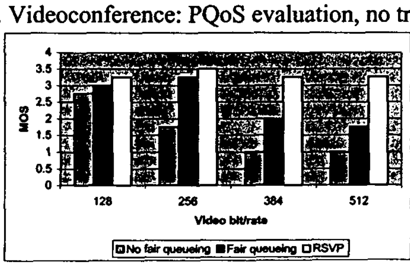

Fig. 3. Video: PQoS evaluation, no transfer.

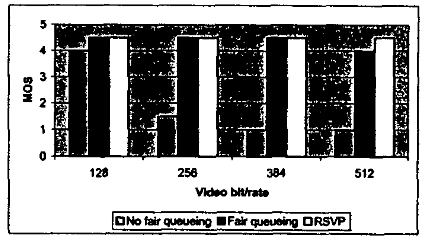

Fig. 4. Voice: PQoS evaluation, no transfer.

The percentage of lost packets in the "No fair queueing" case drastically increases and heavily influences the quality of the comprehension. The overall judgement, in fact, Fig. 2, is very low. The corresponding values concerning video have been shown in Fig. 6. In this case, the lost packets increase also in the "Fair queueing" case and only "RSVP" allows to guarantee high quality (Fig. 3).

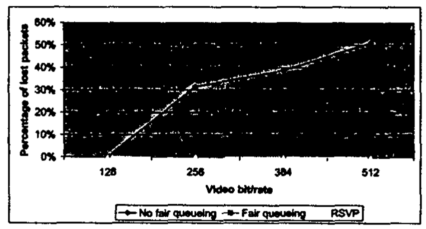

Fig. 5. Voice packet loss, no transfer.

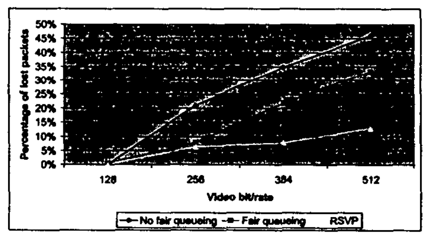

Fig. 6. Video packet loss, no transfer.

Nevertheless, due to the high quality of the audio, the general evaluation (Fig. 2) is not so reduced for the "Fair queueing".

It is interesting to investigate the behavior in presence of an imposed disturbance. Fig. 7, Fig. 8 and Fig. 9 contains the same quantities of Fig. 2, Fig. 3 and Fig. 4 when an FTP transfer (i.e. a TCP-based transfer) is performed. The traffic jam, not individuated neither by the "No fair queueing" nor by the "Fair queueing" algorithm, make the performance worst. Only the results obtained with RSVP maintain a certain level of quality of service. It is important to note the behavior of "Fair queueing": it allows to guarantee bandwidth for the voice flow (Fig. 9) but fails to serve the video, which provides so a very low quality (Fig. 8) to affect the global evaluation (Fig. 7). As in the previous case, the evaluation of the P-QoS may be matched by the objective metrics. Fig. 10 contains the voice packet loss in case of FTP transfer. "RSVP" and "Fair queueing" provide very low losses while "No fair queueing" causes a high packet loss even for low video rate. The percentage of lost packets for video is reported in Fig. 11. In this case, only "RSVP" limits the number of packet lost. It is relevant to point out 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 kbit/s, but two voice flows seldom overlap in a destination since it would correspond to two speakers at the same time), the video flow is completely guaranteed only for a video bit/rate of 128 kbit/s. "RSVP" really guarantees the flows and makes the effect of the disturbance almost negligible.

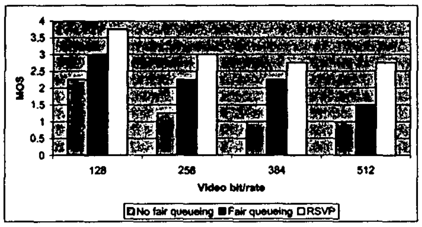

Fig. 7. Videoconference: PQoS evaluation, TCP transfer.

The jamming 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 ("Fair queueing" and "No fair queueing"), where the FTP transfer is treated as the work session flows. The behavior should be clear from Fig. 12 where the average bit/rate of the FTP session is reported for the three schemes considered. Similar considerations may be done if the jamming flow is UDP-based, namely a disturbance video flow.

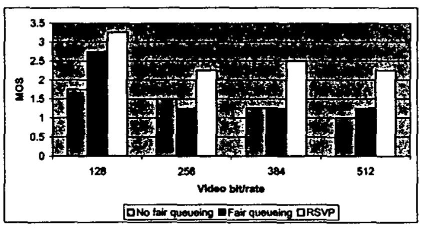

Fig. 8. Video: PQoS evaluation, TCP transfer.

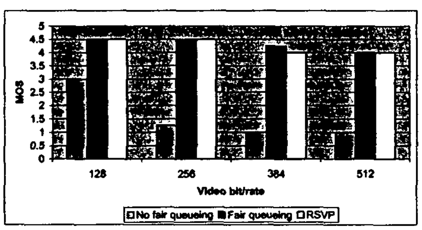

Fig. 9. Voice: PQoS evaluation, TCP transfer.

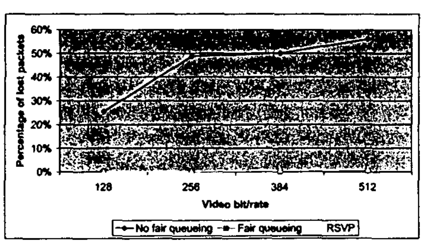

Fig. 10. Voice packet loss, TCP transfer.

Fig. 13, Fig. 14, Fig. 15, Fig. 16 and Fig. 17 contain, respectively, the same quantities as Fig 7. Fig. 8, Fig. 9, Fig. 10, Fig. 11 and Fig. 12, but an UDP-based transfer is imposed in this case.

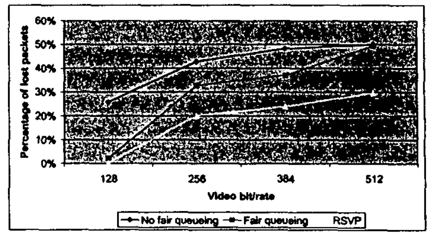

Fig. 11. Video packet loss, TCP transfer.

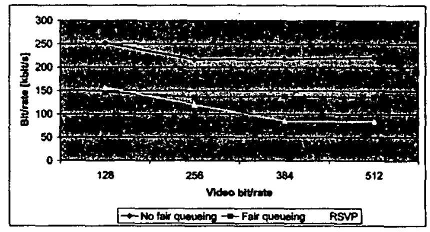

Fig. 12. FTP transfer average bit rate.

The performance is similar to the previous TCP case but the reduction of quality, for "No fair queueing" and "Fair queueing", is even more evident since low video bit/rates (Fig. 13, Fig. 14, Fig. 15 at 128 kbit/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 important flows; the disturbance video flow, which may also be considered as a non guaranteed video flow over the same network, uses only the residual bandwidth.

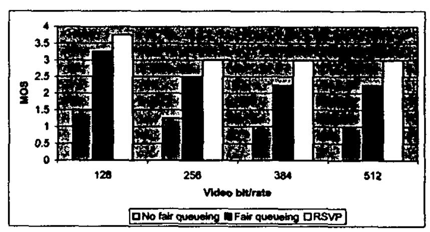

Fig. 13. Videoconference: PQoS evaluation, UDP transfer.

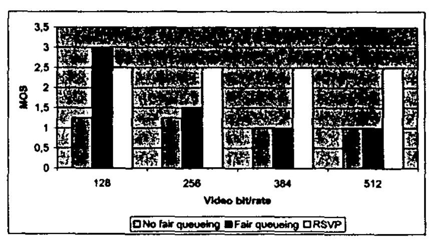

Fig. 14. Video: PQoS evalution, UDP transfer.

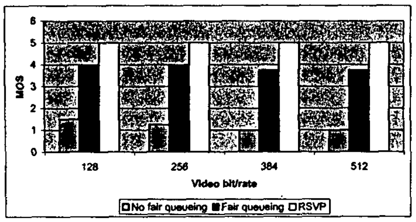

Fig. 15. Voice: PQoS evalution, UDP transfer.

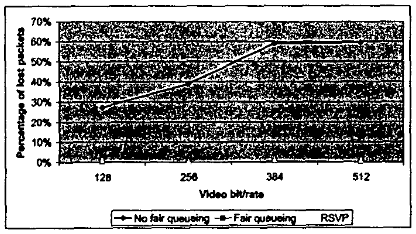

Fig. 16. Voice packet loss, UDP transfer.

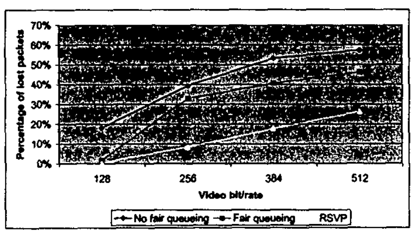

Fig. 17. Video packet loss, UDP transfer.

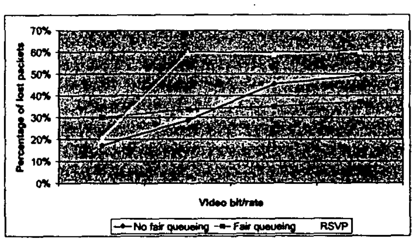

Fig. 18. Disturbance video packet loss.

Its performance, in fact, is really negative, as shown by the "RSVP" results in Fig. 18, where the percentage of dropped packets is reported for the jamming video. On the contrary, the results obtained by the other two algorithms (Fig. 18) are not too different from the same quantities measured for the work session video (Fig. 17). Actually, averaging 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 Genova, reduces the differences imposed by the different transmission medium. For instance, due to the characteristics of satellite channel [4], 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.

VI. Conclusions

Three ways of supporting QoS in a IP satellite environment to be used for videoconferencing and remote distance learning applications have been tested in this paper: a FIFO, a fair queueing and an RSVP - based fair queueing mechanism. 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; the subjective metric has been associated with objective measures as the packet loss rate and the average transmission rate. The matching of the results has allowed to get a better investigation and to completely justify the MOS values obtained.

The results have allowed to get 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.

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