M. Cannarella, M. Draoli, G. Gambosi, C. Gaibisso, M. Lancia

DESIGNING RELIABLE ATM NETWORKS FOR MULTIMEDIA INTERACTIVE IP APPLICATIONS

R. 483 Novembre 1998

Massimo Cannarella – Consultancy and Projects Group, Via della Camilluccia 693, Roma, Italy. Email: m.cannarella@cpg.it.

Mauro Draoli – Servizio Reti di Comunicazione del CNR, P.le Aldo Moro 7 - 00185 Roma, Italy. Email: draoli@src.cnr.it.

Giorgio Gambosi – Dipartimento di Matematica, Università di Roma “Tor Vergata”, Via della Ricerca Scientifica - 00185 Roma, Italy. Email: gambosi@mat.uniroma2.it.

Carlo Gaibisso – Istituto di Analisi dei Sistemi ed Informatica del CNR, Viale Manzoni 30 - 00185 Roma, Italy. Email: gaibisso@iasi.rm.cnr.it.

Maurizio Lancia – Centro di Elaborazione Dati del CNR, P.le Aldo Moro 7 - 00185 Roma, Italy. Email: lancia@iasi.rm.cnr.it.

This work has been presented at the “1998 IEEE ATM Workshop”. May 26-29, 1998. George Mason University, Fairfax, VA, USA.
Abstract

This paper mainly concerns with the design and the performance evaluation of network systems offering a high quality, high reliable videoconferencing service to a community of users dispersed on a metropolitan area. A network topology has been considered in which Ethernet technology based LAN segments are interconnected by a triangular ATM backbone. The main issues of interest in the paper are: the selection and the evaluation of meaningful indices of user satisfaction; the efficiency of Ethernet technology based LAN segments in supporting the real time transmission of multimedia flows; the dimensioning of ATM virtual channels established on a single link; the potentiality of IP and ATM integrated architectures in maintaining the quality of multimedia interactive services in presence of link failure conditions. Several series of simulations have been performed, dealing with different topological configurations, numbers of simultaneously active videoconference sessions, users locations and temporal sequences of activation of the traffic generators. The quality of the user perception has been evaluated with respect to different simulation scenarios. The simulation results show that the ATM technology is a very satisfactory solution to legacy LANs internetworking. ATM link failure tolerance is reached, in a transparent way, simply rerouting ATM switched virtual channels on the available links.
1. Introduction

This paper mainly concerns with the design and the performance evaluation of network systems offering a high quality, high reliable videoconferencing service to a community of users dispersed on a metropolitan area ([1], [2], [3]). We will refer to a network topology in which Ethernet technology based LAN segments are interconnected by a triangular ATM ([4], [5]) backbone and routing/switching devices ([6], [7], [8]). In this context a videoconference service will be considered reliable, if in case of any ATM link failure, the user quality of perception will still be satisfactory. The efficiency of LANs in offering low cost bandwidth to support real time video applications makes their interconnection through a backbone a natural solution to extend broadband services to a metropolitan or geographical area. The choice of Ethernet, aware of the performance constraints introduced by this technology, is due to the need of maintaining user’s investments ([9]). ATM is currently the most attractive solution to the transmission of high interactive multimedia flows, due to its high transfer rate and to the guaranteed QoS of established connections. On the other side, the great diffusion of Internet technologies caused most current multimedia applications to be based on the UDP/IP ([10]) connectionless protocols at the transport and network layers. Unfortunately, integrating the ATM and IP technologies introduces a lot of technological problems, which could dramatically reduce the performance improvements promised by pure ATM networks.

The main issues we deal with in this paper are:

- the selection and the evaluation of meaningful indices of user satisfaction;
- the efficiency of Ethernet technology based LAN segments in supporting the real time transmission of audio and video flows;
- the dimensioning of ATM virtual channels established on a single link. This is really a hard problem to solve in this context, since from the ATM backbone point of view, there is no way to discriminate among incoming different flows (e.g. video and audio), with different QoS requirements.
- the potentiality of IP and ATM integrated architectures in maintaining the quality of multimedia interactive services in presence of link failure conditions. We evaluate the effectiveness, from this point of view, of two architectural configurations which mainly differ on the layer implementing the routing functionality (IP and ATM).

2. The Simulation Scenario

This section illustrates the characteristics of the particular simulation scenario we considered in terms of the adopted network configuration and protocol stack. The whole network system, as shown in figure 1, is composed by a triangular ATM backbone, based onto the standard DS3 transmission system (nominal rate 44.736 Mbps). The backbone can be accessed by multi-port high performance routers, allowing the interconnection of several LAN segments. The network spans over a metropolitan area. Workstations connected to the same LAN segment are supposed to be very close each others (inside the same building). Two LAN segments are connected to the same router and dedicated to the videoconference.

The protocol stack adopted for our simulations is shown in figure 2.

The main network protocol requirements for multimedia applications are low transmission delays and delay jitters. The loss of information, if it does not exceed a critical threshold, is not a crucial matter in this context.
TCP does not effectively support the real time transmission of multimedia data. Its main drawback is represented by packet retransmission, which could result in an additional significant delay and bandwidth occupation and, even worse, in a consistent delay jitter in delivering information. UDP does not retransmit packets. This potentially reduces the transmission delay and the bandwidth occupation. Unfortunately, any UDP entity is a blind source delivering datagrams to the network with no regard to the bandwidth availability. As a consequence if the network is already overloaded, the entity keeps on trying to occupy the available bandwidth, inducing stronger queue occupancy problems, or in the worst case, causing queue congestion and frame loss phenomena. In other words, the delay jitter still cannot be kept under control. IP is mainly in charge of routing datagrams on the basis of their IP address. The ARP layer dynamically translates the IP addresses into Ethernet addresses and vice versa. Ethernet frames are reassembled into IP datagrams by the router, and once again segmented into Ethernet Frames, if they are not going to leave the local area, or into AAL/5 Protocol Data Units, if their destination is reachable through the backbone. The classical IP over ATM layer translates IP addresses into ATM addresses and vice versa. The AAL/5 layers defines the set of services.
provided by ATM. Finally the ATM layer is mainly in charge of establishing virtual channels and virtual paths by which the ATM cells can be moved to their destination.

3. The Modeling Activity

In this section we describe the most relevant steps of our modeling and simulation activities. Firstly we characterize the audio and video streams by which our simulations are fed. We then consider four main subnetwork systems:

- a single LAN segment;
- two LAN segments connected to the same router;
- a single ATM link together with the connected routers and LAN segments;
- the whole system consisting in 3 pairs of LAN segments connected by a triangular ATM backbone.

In order to avoid running complex and lengthy simulations we first of all simulate the behavior of architecturally simpler subsystems. Intermediate results are used to dimension and tune the simulation model of the whole system. They also establish reference values in evaluating the impact of ATM link failures on the quality of multimedia interactive services. All the simulations models have been built by Opnet, a discrete event simulation framework by MIL 3, Inc. ([12]).

The simulations have been finalized to evaluate the effectiveness of the proposed solutions in terms of the quality of the user perception. The average and the 95-percentile end-to-end delay between application entities have been selected as the target evaluation indices for such aspect. The maximum tolerable value of the 95-percentile has been set to 300 msec ([1]) in order to guarantee a good quality of human interaction. We separately measured the audio and video end-to-end delays, in order to evaluate the lip sync possibilities. Several series of simulations have been run for each considered subsystem, with increasing traffic loads. As a result, we identified the values of the traffic descriptors leading to a good quality of the user perception, at the same time, trying to minimize the allocated bandwidth.

Finally, we also take into consideration link failure tolerance requirements and we evaluate the effectiveness of two architectural configurations which mainly differ on the layer implementing the routing functionality (IP and ATM).

3.1. Video and Audio Streams Characterization

As concerns the characterization of the video stream of reference in our simulations, we will refer to the study already presented in [3]. In that paper the authors tested several videconferencing applications on a Sparc Workstation equipped with a SunVideo board (24 bits per pixel). With respect to this particular hardware platform, ShowMe$^TM$ ([14]) resulted to be the application best fitting their needs, which substantially coincide with ours.

ShowMe$^TM$ makes it possible for the user to fix the frame rate from 1 to 30 frames per second and to select among 176x144, 384x288 and 640x480 pixels 256 colors images. The compression algorithm adopted by ShowMe$^TM$ (CellB), is a complex mechanism which implements both intra-frame and inter-frame compression techniques, operating a non reversible transformation of $4 \times 4$ blocks of pixels. In addition, similar corresponding blocks in successive frames are simply skipped. As a consequence, the resulting compressed video frames have a variable length. The authors concentrated their attention on the streams generated by rather still images, as usual in videconferencing sessions, and experimentally investigated their characteristics. Both the average
bit rate, 1.7 Mbps, and the average size of compressed frames, 8634 bytes with a standard deviation of 225 bytes, have been determined.

As far as the end-to-end frame delay is concerned, the SunVideo board does not start compressing a video frame until it has not been entirely buffered. As a consequence, a delay of one frame period, i.e. 40 msec, is accumulated from the time a scene is captured to the time the application makes the corresponding video frame available to the UDP source entity. At the receiver, the delay introduced by the decoder is 40 msec, since it operates on a frame by frame basis too. As a consequence, in order to not exceed the maximum tolerable 95-percentile end-to-end delay, fixed to 300 msec in this context, the difference between the time in which the application source entity makes the frame available to the UDP source entity and that in which the same frame is delivered by the UDP destination entity to the destination application entity must not exceed 220 msec.

The audio stream has been modeled by a statistical traffic generator ([11]). A two state Markov Modulated Poisson Process has been used to model a PCM coded stream with silence detection. The average duration of the silence and activity periods have been respectively fixed to 2.25 and 1.5 seconds. A very easy computation onto the overhead introduced by UDP and IP lead to fix the audio packet size to 64 bytes and the emission rate to 1 packet every 8 msec.

3.2. The Ethernet, UDP and IP Models

The adopted Ethernet model is fully conformed to the IEEE 802.3 ([13]) recommendation for 10BaseT Ethernet segments. Signals are assumed to propagate at 0.66 times the light speed.

The UDP and IP adopted models implement the main functionality of both protocols. Since we assume the reader to be already familiar with their behaviors, only the IP model will be briefly described. Each IP entity is modeled by a FIFO queue system with a fixed service rate of 1,000 packets per second. Such a rate, usually supported by not overloaded workstations, is high enough to have a minor influence on the whole system performance.

Each video frame is fragmented by the IP layer in a sequence of Ethernet frames having a maximum length of 1,500 bytes plus overhead. Since the average size of the transmitted video frames is 8,634 bytes, 5 or 6 Ethernet frames are usually required for the transmission of a single video frame. The loss of any of such frames causes the entire video frame to be lost. Unfortunately, the resulting traffic on the Ethernet bus is extremely bursty.

3.3. The ATM Switching Device Model

The adopted ATM switching device model implements UNI 3.0 and NNI public interfaces on the basis of ITU-T and ATM Forum specifications. Point-to-point, full duplex virtual channel connections are dynamically established and released through suitable signaling.

Virtual path connections are instead established on a permanent basis through manual configuration: a virtual path link has been configured between each pair of routers to support any number of virtual path connections. In the model, the length of such virtual path connections is limited to one virtual path link.

In order to simplify the Call Admission Control, the Usage Parameter Control and the routing procedures, a given QoS has been assigned to each virtual path connection in such a way that only cells belonging to calls in a certain class can be transferred through the corresponding connection.

The link capacity has been defined in accord to the DS3 transmission system standard (44.737 Mbit/sec).
A virtual path connection has been defined on each virtual path link for signaling, including two virtual channels dedicated, respectively, to call management (5.088 Mbit/sec) and to the transfer of routing information (0.932 Mbit/sec).

In what follows, we will use the Maximum Data Rate (MDR) traffic descriptor instead of the standard PCR (Peak Cell Rate) defined by ATM Forum and ITU-T. The MDR is the value, in bit per second, of the PCR, excluding the 5-byte cell header.

Both virtual path connection and virtual channel connection switching are supported. The propagation delay in crossing a switch and the translation delays for virtual path and virtual channel identifiers have been modeled on the basis of real devices performances and set to $2.7 \cdot 10^{-6}$ sec, $10^{-10}$ sec, and $10^{-11}$ sec, respectively.

Moreover, we assumed that virtual path and virtual channel switching is fast enough to support the maximum data rate for incoming cells: this made it necessary to model only output buffers. In particular, a queue is maintained for each class of service and all such queues are served on a Round Robin basis.

All videoconferencing traffic is served in class D and the buffer size at switches for this class is set to 1000 cells.

For what concerns traffic management and control, Call Admission Control and Usage Parameter Control have been implemented.

Routing on the ATM backbone is performed dynamically and in a distributed fashion by using the Bellmann-Ford algorithm. The update interval for routing tables has been set to 120 sec.

The AAL layer implemented in the model is the type 5 one, configured to provide a transfer service for Service Data Units (SDU) in a not assured and block mode way. The AAL takes care of both signaling and data transfer, explicitly modeling segmentation and reassembling operations.

4. The Simulation Activity and Results

Several series of simulations have been run, investigating the average and the 95-percentile end-to-end delays at both the application and the AAL/5 layers. Each series of simulations is mainly characterized by the topological configuration of the investigated subsystem, by the number of simultaneously active videoconference sessions, by the location of the involved workstations and by the temporal sequence of activation of the traffic generators feeding the simulation.

This last aspect, in fact, greatly influences the probability of burst superposition phenomena and consequently the performance of the whole system. The same MDR descriptor has been specified for all the routers in the same simulation, while simulations inside the same series differ by the associated MDR descriptors.

4.1. Local Subsystems

We first of all restrict ourselves to consider local subsystems, i.e. a single LAN segment and two LAN segments connected by the same router. With respect to both these subsystems, the simulation activity has been finalized to evaluate the maximum number of supportable videoconference sessions.

Unfortunately, a single LAN segment effectively supports the transmission of 4 video streams, but does not support 2 simultaneously active videoconference sessions. Audio streams have in fact a dramatic impact on the system performances. This is mainly due, despite of the very
small bandwidth requirements, to the great number of small sized audio packets flowing through the network.

Let us now consider two LAN segments connected by the same router and let us assume each session does not involve users connected to the same LAN segment, as shown in figure 3, where workstations labeled by the same capital letter are involved in the same videoconference session.

Several simulations have been run, which revealed two completely different system behaviors, depending on the temporal sequence of activation of the traffic generators. If they are simultaneously activated the subsystem very poorly performed. On the contrary, if the traffic generator are not synchronized, which is a much more probable event, the subsystem effectively supports two simultaneously active video conference sessions, as shown in figure 4, where the average values of the end-to-end delays collected during the simulations are reported.

Not surprisingly this subsystem better performs than the single LAN segment, since even if 4 video and audio streams are still present on each LAN segment, they are generated by 3 network devices instead of 4.

4.2. Single ATM Link

We next focused our attention on an architecturally more sophisticated scenario in which two LAN segments are connected by a single ATM link. We first of all consider just one active videoconference session over the network, as shown in figure 5.

Two different series of simulations have been run, mainly finalized to investigate the minimum bandwidth that has to be allocated to the virtual channel in order to not exceed the maximum tolerable 95-percentile end-to-end delay at the application layer. The first and the second series deal with not synchronized and synchronized traffic generators, respectively.

Figures 6 and 7, respectively report the results collected by the first and the second series of simulations for increasing values of the bandwidth allocated to the virtual channel. It is worth
noticing that:

- 1.85 Mbit/sec is the bandwidth threshold to obtain a good quality of user perception;
- a main fraction of the whole end-to-end delay is generated by ATM. This is mainly due to the LAN segments under utilization;
- synchronized traffic generators do not obviously induce burst superposition phenomena.

In the next considered scenario, shown in figure 8, we double the number of simultaneously active videoconference sessions. Once again, two different series of simulations have been run, mainly finalized to investigate the minimum bandwidth that has to be allocated to the virtual channel in order to not exceed the maximum tolerable 95-percentile end-to-end delay at the application layer. The first and the second series deal with not synchronized and synchronized traffic generators, respectively.

Figures 9 and 10, respectively report the results collected by the first and the second series of simulations for increasing values of the bandwidth allocated to the virtual channel. It is worth noticing that:

- 3.7 Mbit/sec is the bandwidth threshold to obtain a good quality of user perception;
- a relevant fraction of the whole end-to-end delay is spent trying to access the LAN segments;
- burst superposition phenomena are now present and have a great impact on both the ATM and LAN components of the delay.

Finally, we consider a simulation scenario in which two videoconference sessions are still simultaneously active over the network, but two LAN segments are connected to each ATM router
and just one active workstation is connected to each of them, as shown in figure 11. Two different series of simulations have been run, finalized to investigate the impact of the user location onto the system performances. The first and the second series deal with not synchronized and synchronized traffic generators, respectively.

Figures 12 and 13, respectively report the results collected by the first and the second series of simulations for increasing values of the bandwidth allocated to the virtual channel. It is worth noticing that:

- 3.7 Mbit/sec is still the bandwidth threshold to obtain a good quality of user perception, but the system performance has a great improvement with respect to the last considered scenario. This is due to the LAN segments under utilization.
- a relevant fraction of the whole end-to-end delay is spent trying to access the LAN segments;
- burst superposition phenomena have a great impact on the ATM component of the delay.

### 4.3. Link Failure Scenarios

In this section we investigate the whole system performances. We also take into consideration ATM link failures.

We first of all consider the behavior of the whole system in absence of faults and with respect to different users locations and numbers of simultaneously active videoconference sessions. As expected the system performances do not differ from those of its single ATM link subsystems,
already investigated in the preceding sections. The results of our simulations are consequently omitted.

Let us now take into consideration the impact of a single link failure on the whole system performances. The first scenario we take into consideration is shown by figure 14. Virtual channels have been dimensioned to 1.85 Mbit/sec. If the ATM link of figure 14, connecting router R0 to router R2 fails, a new virtual channel, still dimensioned to 1.85 Mbit/sec, is dynamically established following the only available physical path connecting R0 and R2. By our simulations, we measured the average and the 95-percentile end-to-end delays at the application and AAL/5 layers. The simulation results are extremely close to that for the already investigated not synchronized traffic generators, single link, single session scenario and are consequently omitted. The excellent system behavior is due to the efficiency of the ATM switching process. On the other hand, this fault tolerance mechanism works if and only if at most half of the bandwidth allocated to the virtual paths is allocated to the virtual channels.

We finally investigated a different fault tolerance mechanism: the routing devices are configured as simple IP routers, i.e. their ATM switching capabilities are not exploited. As a consequence 3 pairs of permanent virtual channels have been established between any pair of
routers. Each of such virtual channels utilizes exactly one physical link. If the ATM link of figure 14, connecting router R0 to router R2 fails, router R1 acts as an IP router with rerouting capabilities. Obviously this fault tolerance mechanism requires that the bandwidth allocated to each virtual channel is at least twice the minimum bandwidth required in order to support 2 simultaneously active videoconference sessions (3.7 Mbit/sec).

The results of our simulations are shown in figure 15, where the average and the 95-percentile end-to-end application delays are reported for the best and the worst served users. It is worth noticing that the link failure introduces some unfairness problems. Problems that do not arise if the ATM layer deals with the failures.

The last considered scenario is shown by figure 16. Three videoconference sessions are simultaneously active instead of 2.

The results of our simulations are shown in figure 17, where the average and the 95-percentile end-to-end application delays are reported for the best and the worst served users.

The link failures introduces great unfairness problems. In addition, the results, when compared with those reported in figure 12, show a relevant decline in the system performances, in particular with respect to the 95-percentile end-to-end delays.
### Table 1

<table>
<thead>
<tr>
<th>Synchronized Traffic Generators</th>
<th>MDR (Mbit/sec)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>A</td>
</tr>
<tr>
<td>3.65</td>
<td>3.7</td>
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<table>
<thead>
<tr>
<th>Application End-to-End Delay (msec)</th>
<th>Avg. (msec)</th>
<th>95 perc. (msec)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cong.</td>
<td>55</td>
<td>65</td>
</tr>
<tr>
<td>95 perc. (msec)</td>
<td>88</td>
<td>98</td>
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<table>
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<tr>
<th>AAL/5 End-to-End Delay (msec)</th>
<th>Avg. (msec)</th>
<th>95 perc. (msec)</th>
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<tr>
<td>Cong.</td>
<td>43.5</td>
<td>23</td>
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<tr>
<td>95 perc. (msec)</td>
<td>78.5</td>
<td>40</td>
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</tbody>
</table>

A: Audio; V: Video.

**Figure 13.**

**Figure 14.**

5. Conclusions

Satisfying high quality requirements of multimedia interactive applications is a hard goal to reach in a packet switching environment spanned over a metropolitan or geographical area.

In this paper, we verified that the ATM technology is a very satisfactory solution to legacy LANs internetworking. The available ATM bandwidth can be efficiently allocated at the same time guaranteeing a good level of user perception. ATM link failure tolerance is reached, in a transparent way, simply rerouting ATM switched virtual channels on the available links.

A simpler fault tolerant architectural solution has been investigated too. We verified that IP rerouting over ATM permanent virtual channels is an alternative solution. Nevertheless, our simulation results show some unfairness and performance problems even in a simple link fails scenario. Further investigation is needed to understand the maximum complexity of the network scenario to which such fault tolerant architectural solution can be applied satisfying the strict requirements of reliable multimedia interactive services.
<table>
<thead>
<tr>
<th>Application End-to-End Delay (msec)</th>
<th>A</th>
<th>V</th>
</tr>
</thead>
<tbody>
<tr>
<td>Best Avg. (msec)</td>
<td>43</td>
<td>48</td>
</tr>
<tr>
<td>Worst Avg. (msec)</td>
<td>46</td>
<td>52</td>
</tr>
<tr>
<td>Best 95-perc. (msec)</td>
<td>71</td>
<td>78</td>
</tr>
<tr>
<td>Worst 95-perc. (msec)</td>
<td>74</td>
<td>82</td>
</tr>
</tbody>
</table>

A: Audio; V: Video.

Figure 15.

![Diagram of network with routers, link failure, and LAN sites](image)

Figure 16.

<table>
<thead>
<tr>
<th>Application End-to-End Delay (msec)</th>
<th>A</th>
<th>V</th>
</tr>
</thead>
<tbody>
<tr>
<td>Best Avg. (msec)</td>
<td>41</td>
<td>48</td>
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<tr>
<td>Worst Avg. (msec)</td>
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<tr>
<td>Best 95-perc. (msec)</td>
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<tr>
<td>Worst 95-perc. (msec)</td>
<td>109</td>
<td>113</td>
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</table>

A: Audio; V: Video.

Figure 17.
References


