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PERFORMANCE ANALYSIS OF SERVICE TRANSMISSION IN THE ATM ENVIRONMENT USING A DISTRIBUTED SIMULATOR

S. Koutroubinas, T. Antonakopoulos, C. Stavroulopoulos, and V. Makios*

Abstract

This paper presents a distributed simulator for performance analysis of various applications in the asynchronous transfer mode (ATM) environment. The simulator is used to determine how the quality of a service, either synchronous or asynchronous, is affected when it is transmitted through an ATM network under various traffic conditions. The simulator has been implemented in a distributed Unix environment using network and operating system functions and simulates the three lower layers of B-ISDN. It has been based on the time-driven approach, and is modular and expandable because it uses standard operating system calls for adapting new applications and services to the simulator environment. The paper presents performance analysis results of constant and variable bit rate services, as well as experimental results on compressed and uncompressed image transmissions.

Key Words

ATM, B-ISDN, performance analysis, simulation

1. Introduction

The Broadband Integrated Services Digital Network (B-ISDN) is probably the most promising communication system for supporting the large number of teleservices that will be developed in the near future. These services have different and sometimes yet unknown requirements, which must be supported by the same basic communication method. The B-ISDN uses the asynchronous transfer mode (ATM), which is based on the transmission of constant length cells, to provide the basic communication mechanism; and various adaptation functions are used to adapt the ATM characteristics to the application requirements [1]. According to [2], the services that will be provided by B-ISDN will diversify and will cover new sectors in the residential and business environment. The switched multi-megabit data service (SMDS) will be used extensively for interconnection of high-speed local area networks, and it is considered a necessity for B-ISDN. Video services will be introduced either as switched-access television for residential customers or as video on demand for residential entertainment and business applications (medicine, education, etc.). Multimedia services will cover several different types of media (voice, video, text, graphics, etc.), and they will be successfully introduced in the B-ISDN when the various technical problems introduced by the wide range of traffic characteristics implied by different media are solved.

The analysis of various components of a multimedia service in relation to the B-ISDN communication parameters is the main focus of this work. The analysis of such a system under various traffic conditions, and especially the determination of how these traffic conditions influence the quality of the supported services, is not a trivial task and requires the solution of complex models [3]. The simulation method is one of the methods used to analyze these systems, especially in cases where the probabilistic nature of the problem, in combination with its complexity, makes the use of other modelling techniques impractical. The simulation is based on the development of a system model in a computing environment and the execution of experiments to analyze its performance or to evaluate alternative solutions [4]. The system under analysis can be described as continuous or discrete, using the events evolution, and as deterministic or stochastic, based on the nature of its processes. Usually, the analysis of a communication system is based on its stochastic nature, and discrete time events are used. The simulation can be performed in various timing levels depending on the system complexity, the nature of system execution, the required output statistics, and the like. Synchronous timing is performed in fixed, appropriately chosen time units, and asynchronous timing is based on events.

In this paper, a distributed simulator for the performance analysis of various applications in the ATM environment is described. The simulator has been built using discrete events; its timing has been based on the chronological events at the user side, and the stochastic conditions have been taken into account for the resulting performance analysis. This work is motivated by the need to determine how the basic communication parameters affect the quality of a multimedia service when it is transmitted through an ATM-based network. The simulator, besides providing the required performance analysis results, is mainly intended to simulate real test-sequences in order to observe visually the effect of communication conditions on service quality. Starting from this point, a complete, reliable, and expandable distributed simulator has been developed.

In section 2 we describe the communication environment and highlight the structure of the developed simulator. In section 3 we describe the characteristics of the

* Laboratory of Electromagnetics, Telecommunications and Technology of Information Division, Electrical Engineering Department, University of Patras, 26500 Patras, Greece, e-mail: antonako@ee.upatras.gr

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B-ISDN services and the models of the traffic conditions, and in section 4 we discuss the performance analysis results and give examples of how the quality of video and still-image frames is affected when they are transmitted through an ATM network.

2. The System Model

The B-ISDN architecture is based on the ATM method, which uses fixed-size cells for information transmission and a type of labelled multiplexing scheme (virtual path/virtual channel - VP/VC) for information routing. The ATM is independent of the supported services, and the adaptation of the service requirements to the ATM characteristics is performed by the ATM adaptation layer (AAL). AAL supports multiple protocols for different user requirements, and these protocols are classified using the timing relation between source and destination, the constant or variable bit rate coded data, and the service connection mode of operation. AAL type 1 is used for supporting synchronous service of constant bit rate, type 2 is for variable bit rate synchronous services, type 3 is for connection-oriented data transfers, and connectionless data transfers are supported by type 4. The AAL is also divided in two sublayers: the convergence sublayer (CS), which is mainly used to satisfy the user interface requirements, and the segmentation and reassembly (SAR) sublayer, which is used for transforming information between protocol data units (PDUs) and ATM cells payload (information field). This layered architecture is used simulating the service support in the developed simulator, and it is shown in fig. 1.

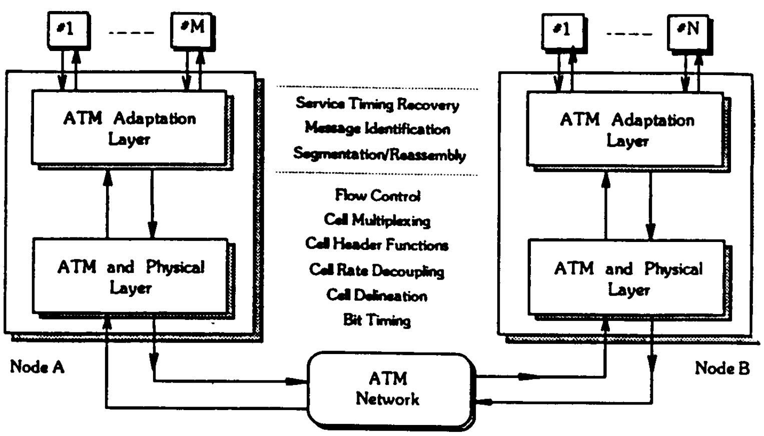


Figure 1. The layered architecture for service support in the ATM environment.

The simulator has been implemented in a distributed environment of three workstations (HP9000/300 series) running over the Unix operating system. The communication between the various processes on the same or different workstation is achieved using network and operating system functions. The simulator is based on the time-driven approach [5], where the simulation clock is incremented at the end of each transmitted cell (fixed length time interval). The simulator is modular and expandable in order to allow easy adaptation of various applications in this test-bed. Its modular structure allows the use of workstations with various capabilities, and the simulation load is balanced according to their performance.

The simulated system consists of four basic levels, which represent the three lower layers of the B-ISDN and the supported service. The simulator has been implemented by using various tasks in a pipeline structure for each information flow, but the execution of tasks of parallel connections or in the same layer is performed using a time-sharing method, since each layer is implemented using a single processing unit. The distribution of the processing power to the various modules of a layer is performed using a scheduler, and its parameters can be easily adapted to the values of a real system. The tasks are organized in four basic modules having the following characteristics:

- *The network unit* simulates the performance of the network part, is independent of the source and the destination nodes, and generates variable delays on the cells of the same connection. This unit is also used to generate a specific cell error rate for each cell stream.
- *The transmitting unit* receives the data from the supported service and implements the adaptation and ATM functions of a node. The AAL part of this unit depends on the type of supported service. When different AAL functions have to be implemented, this can be easily done by modifying the system configuration file. Owing to its modular structure, the software determines its functionality during the program start-up, and only the necessary functions and modules are included. The transmitting unit has one ATM and multiple AAL entities, and it can be configured to support multiple bit streams with different quality of service characteristics.
- *The receiving unit* collects data from the network and regenerates the user information transmitted through the network. Its structure is the same as that of the transmitting unit and performs the decoding and demultiplexing procedures. Each network node can be emulated using various modules of transmitting and receiving units, according to the requirements of the supported services.
- *The supported service unit* is used to generate real traffic using various traffic models. The generated data can be either "dump" data or "real" data. Dump data are random data used only for collecting system statistics and measurements, and real data (like video frames, audio sequences, etc.) are used for visual or acoustical perception of the communication system influence. The user data sources and sinks are used for inserting load into the network, for presenting the simulation results and images for further evaluation, and for handling the file system of the simulator, where data are stored and retrieved. Because the simulator uses the cell transmission time as the system time unit, the user data interface uses or generates the timing references for storing and retrieving synchronous services.

The simulator performs all the functions (segmentation, assembly, cell header generation, etc.) required by the network, and its elaborate model description guarantees the correctness of the simulator. In fig. 2, the simulator's model architecture is shown. The AAL is used to implement the functions of each service class for achieving the required QOS. Although four service classes have been explicitly defined, two different adaptation functions have been implemented in order to support constant and variable bit rate of connection-oriented services. No adaptation for connectionless type services has been supported.

they are described by their peak rates. A CBR source is considered active during a connection, and no silent periods exist during the connection. The constant bit rate is achieved by varying the source coding algorithm. That results either in waste of bandwidth during periods of small information content or in quality degradation for periods of large information content. Especially in CBR video, a smoothing buffer is used because the information content

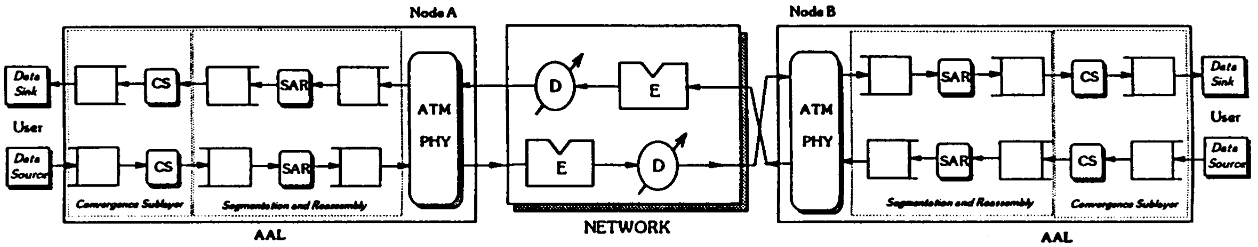


Figure 2. The simulator model.

differs from one frame to the other.

Constant bit rate services (CBR), such as audio, video, and telephony, are supported by AAL type 1. The received service data units are segmented to form 48-byte long ATM cell payload using a byte for sequence numbering and protection. Two synchronization methods have been implemented, using either implicit timing, by monitoring the buffer filling level, or explicit timing via time stamps.

Variable bit rate services (VBR), such as compressed video, are simulated as streaming mode services [6]. Variable length service data units are used by the convergence sublayer for padding and header/trailer insertion, and then the segmentation is performed for generating ATM cell payloads. An internal pipelining function has been implemented in the AAL type 2 transmitting side in order to initiate transfers to the receiving entity before completion of the SDU reception. The AAL type 2 supports nonassured operations as no lost or corrupted SDUs are corrected by retransmission. The specific functions used in each submodule (delay variation, available bandwidth, etc.) are described in the next section.

3. Service and Traffic Models

The ATM environment supports services with constant and variable bit rates as well as connection-oriented and connectionless transmissions. The services have been classified by CCITT [7] as conversational, retrieval, messaging, and distribution.

Although extensive work has already been done in analyzing various aspects of communication networks, the behaviour of data sources is not well understood, and for a given class of data services there is no typical source behaviour. The characterization of a source is performed by defining its traffic characteristics and its quality-of-service (QOS) requirements. The QOS metrics are categorized as either call control or information transfer parameters. Because the CBR services generate traffic at a constant rate,

The main benefit of ATM is exploited when variable bit rate (VBR) services are used, because traffic statistical multiplexing is performed. In VBR services, the source coding algorithm preserves the service quality, but the resulting bit rate varies with the time and its peak-to-average bit rate is greater than one. There are various voice source models that assume that the duration for active and silent periods is exponentially distributed [8]. Such an interrupted Poisson process (IPP) results to a packet generation process, which has its two first moments equal to:

$$\text{mean value} = \frac{\sigma_A + \sigma_s}{\lambda \cdot \sigma_s} \tag{1}$$

and

$$\text{coefficient variation} = \text{sqrt}\left[1 + \frac{2 \cdot \lambda \cdot \sigma_A}{(\sigma_A + \sigma_s)^2}\right] \tag{2}$$

where $1/\sigma_A$ and $1/\sigma_s$ are the average duration of the active and silent periods and λ is the packet generation rate during the active period. The use of slotted time, as slot is considered the ATM cell duration, results in voice coding as interrupted Bernoulli process. The simulator has been used for measuring conversational services characteristics. For storing data of real conversations we used two HP9000/700 workstations because they have audio capabilities. The data were then analyzed using the distributed simulator. As the experimental results showed, the above mathematical formulas, in most of the cases, can be used for modelling the voice sources.

Various models have been proposed for describing a CBR service. The simplest model assumes that a byte is stored in the user-AAL buffer in constant time intervals, whereas more realistic models assume that a burst is generated periodically and each burst contains a constant number of bytes. At the receiving node, the CBR service reads a number of bytes periodically from the output buffer. During normal system operation, no data are lost owing to the network capacity limitations and traffic statistical multiplexing. In this case the relation between the traffic statistics, the service generation rate, the AAL-PDU length, and the AAL processing times can be analyzed.

When a VBR service is considered, new problems arise in the service modelling. As was explained previously, variable bit rate is generated when constant image quality is pursued and the generated bit rate is adapted to the local and temporal image complexity. Various results concerning source modelling for video teleconferencing services have been presented in [6], and they have been used as the basic information for our models. Each VBR service is characterized by its mean and peak bit rate and the data are stored in the user-AAL buffer using the peak rate. The VBR service generates bursts of data in constant time intervals (like the interframe period of the video system), but the burst length varies depending on the frame complexity. When a VBR service is analyzed in this simulator, the length of its bursts follows either a uniform distribution or a gamma distribution.

During the connection set-up procedure of a synchronous service, a part of the network bandwidth is allocated for use by the specific service, and the existence of unassigned cells is used by the ATM layer to determine if there is available bandwidth for a specific AAL. The available bandwidth varies depending on the statistical multiplexing performed at the network owing to the variable bit rate nature of the total traffic. For these reasons a Poisson distribution with lower and upper limits has been used to describe the available cells for data transmission.

When the cells of a connection pass through the network, they are subject to errors (transmission errors, switching buffer overflow) and to different delays. The cells of a connection follow the same VP/VC routing and no change to their order of arrival can be performed. The delay functions at the network unit have also been implemented using Poisson distribution, and for the cell error function a uniform distribution has been used. The Poisson process has been selected for modelling the delay functions because the sum of a large number of independent stationary renewal processes (that is the case for the network offered traffic that determines the network delay) tends to a Poisson process [9]. The error function follows the uniform process because the errors are mainly due to transmission conditions and are uniformly distributed.

4. Experimental Results

The developed simulator was used to analyze primarily synchronous services in the ATM environment. In this section some representative results are discussed.

4.1 Picture Quality

Although the simulator was mainly used for analyzing video and image compression algorithms, these results are not easily presented in text. In this section two examples are presented. The first concerns uncompressed video transmission and the second JPEG still-picture compression.

For uncompressed video, a number of video frames have been transmitted through the network under various traffic conditions. In fig. 3(a) the original "smiling technician" frame is shown. This frame has been transmitted using either (a) uncompressed, no line synchronization format or (b) uncompressed, start of line control format. Fig. 3(b) shows how the cell error rate, which occurred because of network statistical multiplexing, affects the picture quality. When no line synchronization is used, a burst of errors makes the frame pixels wrap around and the rest of the frame is destroyed even when there are no more transmission errors. When a line resynchronization method is used (fig. 3(c)) the occurrence of a burst of errors affects only the specific line and resynchronization is achieved in the next line.

The JPEG compression standard for continuous-tone still images [10] was designed for compressing "natural" scenes and follows a "lossy" scheme by exploiting known limitations of the human eye. Although JPEG compressed

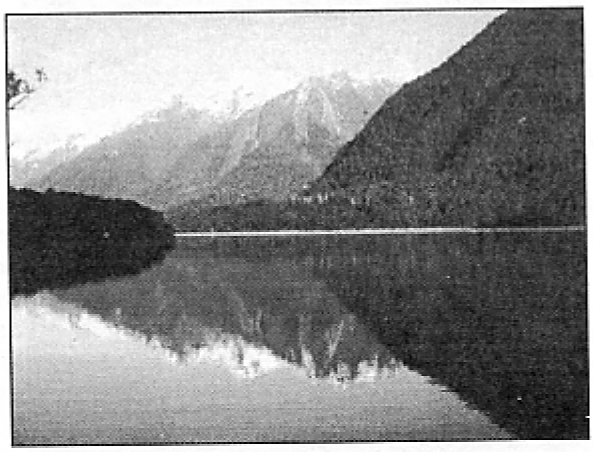
(a)

(b)

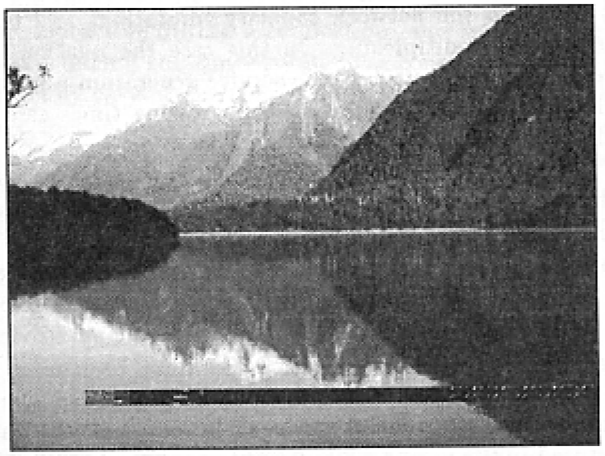
(c)

Figure 3. Video frames transmitted through the ATM simulator: (a) the original frame; (b) errors in uncompressed frame without synchronization characters; (c) errors in uncompressed frame with synchronization characters at the beginning of each line.

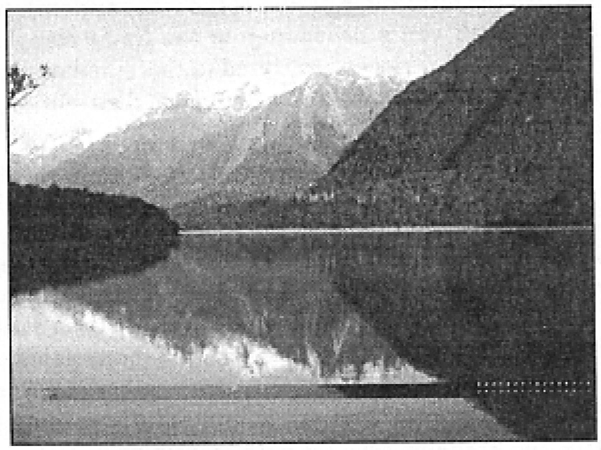
images can be transmitted as files and backward-error correction mechanisms can be used, in this example the JPEG coder output was processed by AAL type 1. Fig. 4 shows the original and the regenerated images after transmission. The original image (fig. 4(a)) was gray-scale, its resolution was 640x400, and it was compressed to 22 kbytes (10:1 ratio). Fig. 4(b) shows the result of a single byte error. Since the image was compressed in blocks, only one block has been affected and synchronization is achieved in the next block that contains a start of line. Figs. 4(c) and (d) show cases where a cell payload has been lost. In the first case an all-zero cell payload is generated for substituting the lost cell, and in the second case the lost cell is just discarded. In both cases, the inserted noise depends on the contained information at this specific image part.



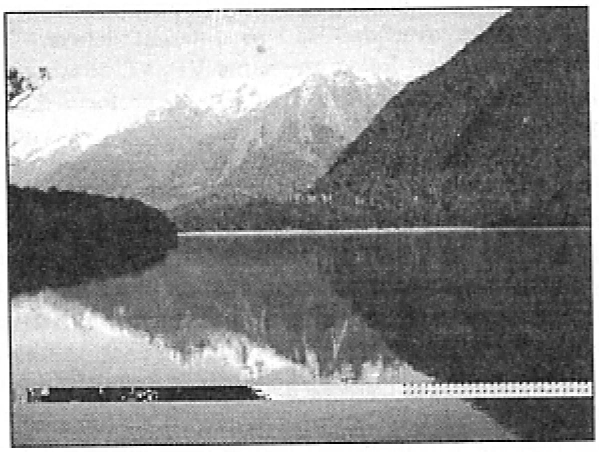
(a)



(b)



(c)

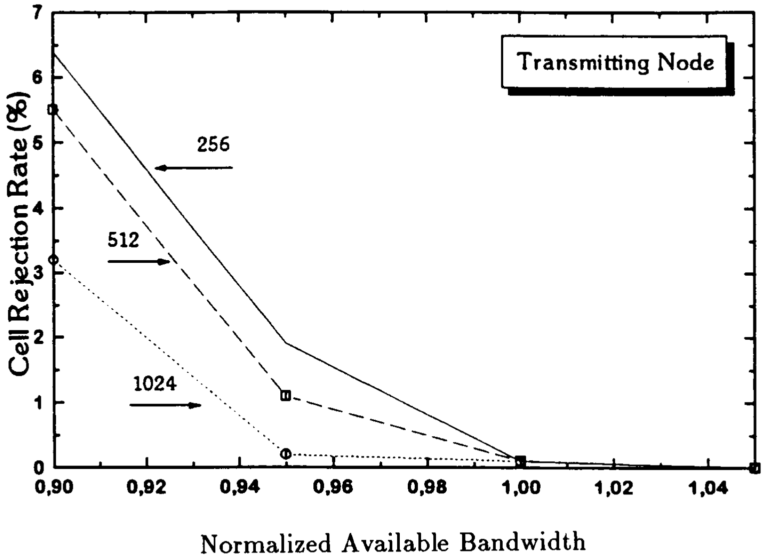


(d)

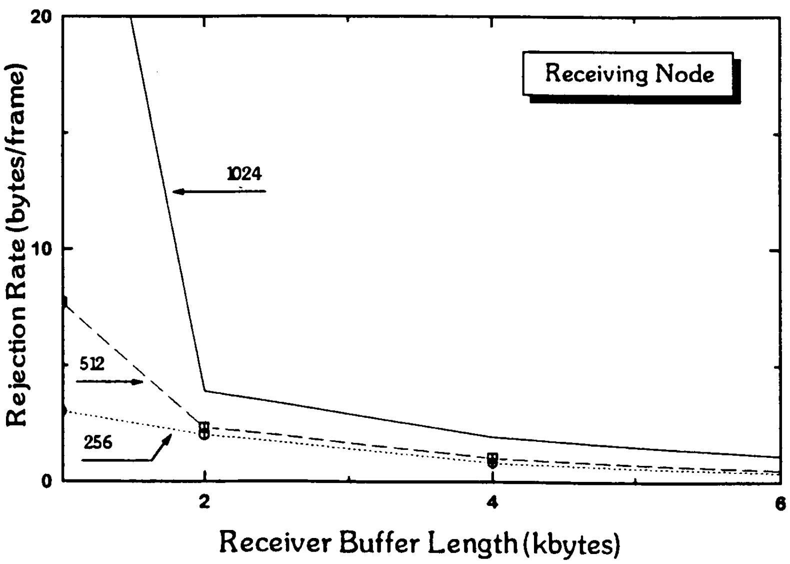
Figure 4. JPEG coded image frames transmitted through the ATM simulator: (a) the original frame; (b) one-byte error in a cell payload; (c) an erroneous cell payload has been substituted with an all-zero payload; (d) a cell payload has been lost and has not been substituted with dump data.

4.2 System Parameters

The simulator has also been used to measure how the AAL performance is affected by some system parameters and how this is related to the type of supported service. We have considered a CBR and a VBR service with similar characteristics, and have studied the influence of the CS-PDU length to the cell rejection rate at the transmitter and the buffer overflow at the receiver. In the transmitting unit, a new burst is received at constant time intervals, and its length is constant for the CBR service and variable for the VBR service, depending on the image complexity. When a new burst is received, the CS buffer is cleared and its content is rejected as its playout time has expired. As is shown in fig. 5(a), the longer the PDU, the smaller the cell rejection rate, because the variations at the network traffic are better smoothened. As the number of sources that contribute to the traffic multiplexing increases, shorter bursts are more easily handled by the network and the rejection rate decreases. Moreover, the burst length variations are not the major factor affecting the statistical bandwidth of



(a)



(b)

Figure 5. The CS-PDU length as a parameter of (a) the cell rejection rate at the transmitting node and (b) the buffer overflow at the receiving node.

the connection; it mainly depends on the characteristics of existing connections in the network. As the experimental results show, the longer the burst length generated at the AAL level, the better the statistical multiplexing that is performed at the network and the better the support of VBR services. The use of position synchronization characters is a must in video transmission because the influence of transmission error is limited only in the area between two consecutive synchronization characters.

At the receiving end, as the PDU length increases, the buffer overflow probability increases when no flow control mechanism is used, because of the increased probability of the arrival of a long burst at the output buffer. The analysis of the same parameters for a VBR service has shown variations in its mean values, depending on the statistical relation between the burst length and the traffic multiplexing. The cell error rate mainly depends on the probability that a cell arriving at a switch finds no empty space and is discarded. Considering the network conditions almost stable due to the statistical multiplexing, the probability of finding the buffer of a switch full is also constant and the cell error can be modelled using a uniform distribution.

5. Conclusions

A distributed simulator for analyzing various applications in the ATM environment was described in this paper. The simulator was based on the time-driven approach and used to determine how the quality of a synchronous service is affected when it is transmitted through an ATM network under various traffic conditions. Experimental results on the analysis of constant and variable bit rate services in the ATM environment were presented, and the influence of various network parameters to the service quality was examined. The proper selection of the length of PDUs send to AAL and the memory management internal to AAL determine the quality disturbances observed when a video stream is transmitted through the network. In order to minimize the synchronization problems due to transmission errors, special protection procedures have to be used for cells carrying timing information.

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Biographies

Stelios Koutroubinas was born in Ioannina, Greece in 1965. He received the Engineering Diploma degree in 1989 from the School of Electrical Engineering at the University of Patras, Patras, Greece. In November 1989, he joined the Laboratory of Electromagnetics at the University of Patras in R&D projects for the Greek Government and the European Economic Community, as a research staff member. At the same time, he started his Ph.D. on Video Mail services. He has 5 publications in the above subject and is actively participating in several ESPRIT and RACE projects of EEC. Mr. Koutroubinas is a member of the Technical Chamber of Greece.

Theodore Antonakopoulos was born in Patras, Greece in 1962. He received the Engineering Diploma degree in 1985, and his Ph.D. degree in 1989 from the School of Electrical Engineering at the University of Patras, Patras, Greece. In September 1985, he joined the Laboratory of Electromagnetics at the University of Patras in R&D projects for the Greek Government and the European Economic Community, initially as a research staff member and subsequently as the senior researcher of the Communications Group. Since 1991 he has been on the faculty of the Electrical Engineering Department at the University of Patras, where he is currently a Lecturer. His research interests are in the areas of data communication networks, LANs, MANs, B-ISDN and wireless networks, with emphasis on efficient hardware implementation and rapid prototyping. He has over 35 publications in the above areas and is actively participating in several ESPRIT and RACE projects of the EEC. Dr. Antonakopoulos serves in the Program Committee of the IEEE International Workshop on Rapid System Prototyping, is a member of the Communications and Computer Societies of the IEEE, and a member of the Technical Chamber of Greece.

Christos Stavroulopoulos was born in Larissa, Greece in 1965. He received the Engineering Diploma degree in 1989 from the School of Electrical Engineering at the University of Patras, Patras, Greece. In November 1989, he joined the Laboratory of Electromagnetics at the University of Patras in R&D projects for the Greek Government and the European Economic Community, as a research staff member. At the same time, he started his Ph.D. on Diagnostics Systems based on Fuzzy Logic. He has 2 publications in the above subject and is actively participating in several ESPRIT and RACE projects of EEC. Mr. Stavroulopoulos is a member of the Technical Chamber of Greece.

Vassilios Makios was born in Kavala, Greece. He received his Electrical Engineering degree (Dipl. Ing.) from the Technical University in Munich, Germany in 1962 and his Ph.D. degree (Dr. Ing.) from the Max Planck Institute for Plasmaphysics and the Technical University in Munich in 1966. From 1962-67 he was a Research Associate in the Max Planck Institute for Plasmaphisics in Munich, where he was associated with microwave interaction studies on plasmas. He served as Assistant Professor in 1967-70, Associate Professor in 1970-73 and Full Professor in 1973-77 in the Department of Electronics, Carleton University in Ottawa, Canada, where he was involved with teaching and research in microwave and optical communications, radar technology, remote sensing and CO_2 laser development. From 1977 he is an honorary Research Professor of Carleton University. Since 1976 he has been Professor of Engineering and Director of the Electromagnetics Laboratory in the Electrical Engineering Department of the University of Patras in Greece, where he is involved in teaching and research in microwave and optical communications, data communications networks, LAN's, MAN's, and B-ISDN with emphasis on efficient hardware implementations and rapid prototyping. He is also involved in research in photovoltaic systems. He has published over 120 papers and holds numerous patents in the above fields. He has participated in the organizing committees of numerous IEEE and European Conferences and was the Technical Program Chairman of the 5th Photovoltaic European Community Conference in Athens 1983 and Co-Chairman of the EURINFO 1988 Conference of the European Community. He is the recipient of the silver medal of the German Electrical Engineering Society (VDE). He is a senior member of the IEEE, member of the Canadian Association of Physicists, the German Physical Society and the VDE, Professional Engineer of the Province of Ontario and the Greek Technical Chamber.