Technical Evaluation System for Telemedicine-Based New Healthcare Services

I.Martínez[1], J. Fernández[1], J. García[1]
[1]Communications Technologies Group (GTC), Aragon Institute for Engineering Research (I3A), University of Zaragoza, Spain

***Abstract*—**The large development of multimedia clinical applications and the use of inter and intra-hospital communication networks imply particular analysis for each one. The study and evaluation of new health services based on telemedicine requires a technical methodology. In this paper we propose a complete methodology implemented by an automated process that includes the system and network modelling and a test prototype for real and simulated measurements. The proposed method can be used to adjust traffic parameters (e.g. transferred clinical file sizes or signal transmission rates), to guarantee desirable Quality of Service (QoS) requirements, under different network resources.

Index Terms—Telemedicine services, technical evaluation, network design, multimedia communication.

I. Introduction

Telecommunications and advanced information technologies have increasingly been used for clinical activities and research to improve health care delivery. With this growth in use, these technologies have undergone many investigations to evaluate their effectiveness, efficiency, and feasibility [1-3]. This technical evaluation requires firstly a measurement methodology [4, 5] in order to analyse applications and networks requirements and their consequently service design and modelling.

Service design is considered by diverse authors as definitive for the correct implementation, performance and maintenance. This design process follows a logical order: first, to determine the available resources in the network, then to analyse the type, volume and QoS requirements of the information to be transferred, and finally to tune applications that the network is going to support. Service modelling is at the heart of any performance evaluation of telecommunications network [6].

Telemedicine services are usually based on multimedia technologies and they are expected to support multiple and diverse clinical applications and networks topologies. Such heterogeneous environments require that different applications should be provided different QoS requirements to accommodate their distinct service types [7]. An accurate estimation of network performance is critical for the success of these multimedia services [5,8].

The Internet Engineering Task Force (IETF) is currently studying models and architectures for supporting differentiated services, in order to better evaluate multimedia streams QoS requirements [2-3,9]. A much extended idea consists on trying to adapt the applications to the network characteristics. A number of experiments have already shed the light on the benefits of adaptation techniques [10,11]. By handling a variety of quality measures, such as packet dropping rate [12], delay jitter [13], bandwidth [14] and variable source rate availability, they allow to improve the quality of e-health communications over best-effort networks [15,16].

In this article, a methodology for technical evaluation of QoS traffic requirements in telemedicine services is proposed. Section II defines the test methodology that includes service definition, service modelling and measurement module. This tool aims at optimizing application design and infering network tomography. Thus, an overview of the telemedicine service design based on different medical routines and communication technologies is given in section III. Section IV defines the complete process of modelling and the evaluation prototype, that include real and simulated measurement tool, is presented in section V. Results obtained from several representative examples of telemedicine services, based on different technological environments, are discussed in section VI.

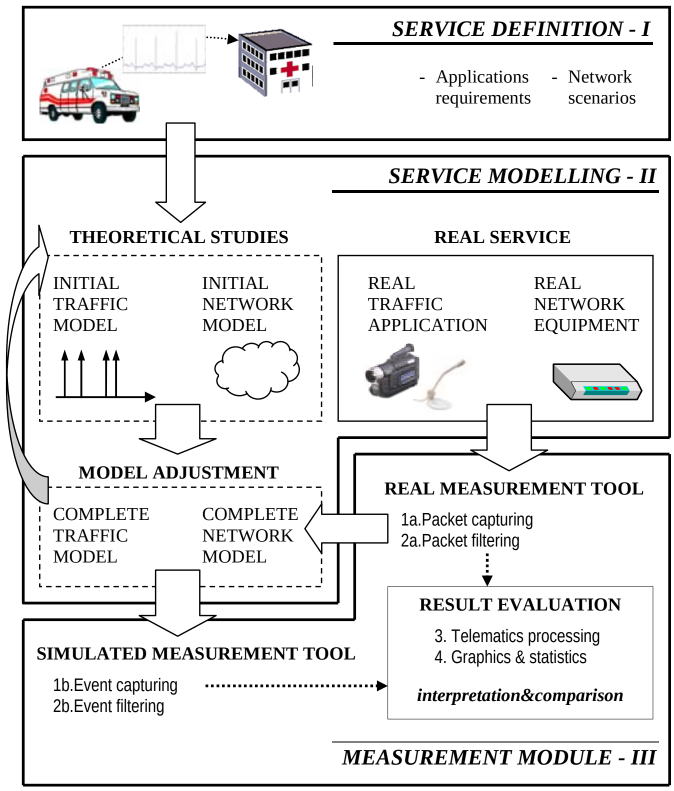


Fig.1. Test methodology basic scheme.

I. Martínez, J. García and J. Fernández is with the Communications Technologies Group (GTC), Aragon Institute for Engineering Research (I3A), University of Zaragoza, Spain.
This work was supported by projects from Comisión Interministerial de Ciencia y Tecnología (CICYT) and Fondos Europeos de Desarrollo Regional (FEDER) TIC2001-2481 and TIC2001-2167-C02:02, and Fondos de Investigación Sanitaria (FIS) FISG03/117.

II. METHODOLOGY

A telemedicine service is usually based on the use of an application under a communications network. We propose a technical evaluation methodology for telemedicine services considering the following steps (see Fig.1):

i. Service Definition. It includes the selection of:

- Application requirements: type, volume and QoS thresholds of the related traffic flow.
- Network scenarios for inter/intra-hospital connections, corresponding to WAN/LAN (Wide/Local Area Network) environments, respectively.

ii. Service Modelling. Firstly, theoretical studies allow translating the previous definitions to an initial model in order to test all the implementation possibilities. Subsequently, from captures of a real service, the complete model should be adequately adjusted to reality considering following items:

- Traffic model, data traces defined by theoretical characteristics adjusted according to captured real traffic application.
- Network model, simulated conditions defined by theoretical characterization adjusted according to captures of real network equipment behaviour.

This adjustment process is iterative (each goal improves theoretical information for further models) and combined (both network models and network devices can be training with real or modelled input traffic flow).

iii. Measurements Module. It integrates the following tools:

- Real measurement tool. It consists on the traffic captures through network devices that allow the experimental characterization of the complete service.
- Simulated measurement tool. It implies the use of the traffic data traces and the network model to obtain time events and simulation results. Simulation creates a model, which is used to explain system behaviour and to see how the system performs under varying conditions to design the service with desirable performance characteristics. Thus, it allows studying a system in well-known conditions where repeatability is necessary in order to understand events much better.
- Result evaluation. Finally, after a telematics processing to obtain suitable traffic parameters (bandwidth, delays, inter-packet times, and so on), the comparison between real and simulated results is necessary to optimize complete service model.

This technical evaluation methodology yields the service optimization. It means being able to infer improvements in the design of application from obtained results. This fact is being a basic key design in telemedicine services in order to allow traffic adaptations (flow rates, compression ratios, and so on) to changing system resources, heterogeneous and non-completely known networks, and consequently, to make a efficiently use of them.

III. SERVICE DEFINITION

A. Application Requirements.

The application of telematics technologies to the sanitary environment implies that characteristics of telemedicine services are associated to multimedia communications. In this article, most representative kinds of medical activities have been considered (see Fig. 2):

- Clinical data transfer: *off-line* transmission of medical tests and administrative files (from medical data exchange between centres or speciality sections).
- Medical consults: patient reports and clinical routine consults that occur during accesses to databases, queries to medical report warehouse, and so on.
- Videoconference: multimedia inter-connection including audio and video between hospitals, mobile units, etc.
- Biomedical signals transfer: *on-line* transmission of vital parameters such as ECG (electrocardiogram) signal, blood pressure or oxygen saturation.

B. Network Scenarios.

The inter/intra hospital communications design is based on the health system hierarchical structure [7].

Studies of WAN environments correspond to network technologies, devices, topologies and, in summary, desirable resources to guarantee communications among geographically dispersed centres and regions. In Fig.2 an inter-communication edge router is shown that concentrates traffic from other health network nodes (as hospitals, reference centres or medical specialities centres) or from mobile units, in a global way.

Moreover, LAN access design corresponds to intra-connection among hospital equipments, signal acquisition cards, information nodes and, in summary, all application generators to study traffic flow distribution. In Fig.2 a scheme of primary health centre model is shown that includes different equipments and utilities distinguishing into fixed (off-line and on-line) and mobile accesses connected to described edge router.

Some representative scenarios have been considered in this article, such as LAN accesses based on switched Ethernet topologies and WAN links based on FR (Frame Relay), ADSL (Asymmetric Digital Subscriber Line) and UMTS (Universal Mobile Telephone Service) technologies, where maximum link-capability, packet dropped rate and delay threshold were analysed.

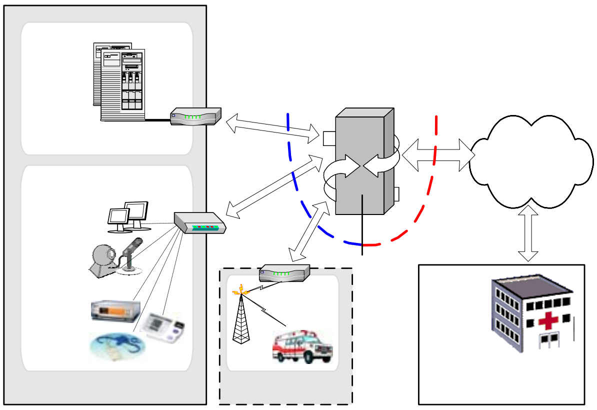


Fig.2. Telemedicine service model.

IV. SERVICE MODELLING

A. Traffic Model.

The telemedicine application model has been derived from two perspectives:

- Real traffic application, including hospital videoconference, biomedical signals and clinical data transfer, database access, etc. This real traffic flow is further used to be sent to the network devices and to train the simulated scenarios.
- Theoretical studies, from the analysis of telemedicine application characteristics (data traces) and from the review of publications [17-19]. Thus, several traffic models have been considered:
  - Non Real Time (NRT) service for *elastic* applications, based on an ASAP (as-soon-as-possible) model. It is basically defined over Transmission Control Protocol (TCP), for non-loss mode transmission, to provide with a connection characterised by dedicated bandwidth, parameterised by a Peak Data Rate (PDR).
  - Real Time (RT) service for *non-elastic* applications (that are "no waiting data arrival" later than a threshold time), based on a multimedia connection model. It is basically defined over User Datagram Protocol (UDP), for non-flow-control transmission, to provide with a connection characterised by low delay and small nonzero random loss ratio, parameterised by a PDR and a Sustained Data Rate (SDR).

B. Network Model.

The network equipment model has been derived from two perspectives:

- Real network equipment, from considered real environment, including interconnection devices (hubs, switches, routers) and WAN links based on FR, ADSL and UMTS technologies (with thresholds of BW< 2Mbps and CLR < 1-3%).
- Theoretical studies, from the analysis of experimental information (laboratory parameters) through a simulated scenario, including restrictive conditions to study *bottleneck*, and from the review of publications [17-19]. The studies include queue size (Q_i) design, scheduling algorithms (based on priority allocation methods), flow control methods, transmission delay (τ_i) thresholds, loss probability models, etc.

The variation ranges of simulation parameters are summarized in Table 1. Results obtained from the simulations are presented and compared with the analytical results in section VI.

Telemed services	parameter	ranging value
Medical test off-line transfer	NRT-ftp	*MTU* = 0.7,1.7,3.5-96MB r_i = 480,540Kbps,1Mbps
Clinical consults (web connection)	NRT-pareto NRT-expo	s_i = 10-20KB Δt_i = 10,40,120s
Biomedical signal *on-line* transmission	RT-cbr	s_i from 80 to 1024B Δt_i from 20,40 to 120ms
Medical videoconference - Audio (VoIP) - Video (ISDN/POTS)	 RT-on/off (G.722/G.728) RT-vbr (H.261/H.263)	 s_i from 240 to 1472B Δt_i from 42 to 126ms

Table 1. Values of simulation parameters.

V. MEASUREMENTS MODULE

Following the defined evaluation methodology (see Fig.1), an integrated measurements module that includes both real and simulated measurement tools has been developed.

A. Real Measurements Tool.

It consists on a C++ program that implements the process of traffic measurement (based on *shell* codes, C processing codes and *tcpdump* [13] software running with *libcap* packet capture library, which uses Berkeley Packet Filter (BPF) system to capture cells of traffic from both network ends). A screen capture of this application is in Fig 3(a). It allows the automatization of the measurement process from different tests (*T*). Each test establishes a network scenario which consists of several connections that are defined by using a configuration input file (*Treal.in*) containing IP addresses and ports for traffic generator and receiver of each connection, and measurement machine. Since input traffic can be provided from a real application or from a trace generator model, this input file also includes the parameters of this related generated traffic.

Once network scenario is defined and implemented, the measurement process is launched, following the RFC-2330 [14] of IPPM (IP Performance Metric) that establishes following steps:

1. Packet Capturing. The application traffic comes from real flow or data traces. In the data traces case, application launches traffic generators in sending and receiving machines and provides them parameters that allow traffic modelling. Using software *tcpdump*, the application traffic is captured by means of the temporal stamp (*timestamp*) that *libcap* applies to each package, following ASAP assignation from *kernel*. Traffic capture can be made in a machine which has two network interfaces or in two different machines. In the first case, since clock reference is the same for both *tcpdump* processes launched in each test point, end-to-end synchronization is guaranteed. In the second case, these processes are synchronized using NTP (Network Time Protocol) commands. As a result of this step two event output files (*Treal.snd/.rcv*, in sending and receiving ends, respectively) are created.
2. Packet Filtering. Using output files from the previous step, packet filtering is made considering sending port and traffic type. As a result of the filtering process, corresponding two files (*Treal-IP-PORT.snd/.rcv*) are created by each defined connection. Each pair of files is analyzed to obtain information referring to captured packets: packet identifier, sending time, reception time, packet size, delay, etc. These results are stored in a event output file (*Treal.res*) in a suitable format.

Each process step can be run in a different machine to obtain parallelism. Command execution in remote machines is made using SSH that provides authentication and safe communication on unsafe networks by means of data codification.

B. Simulated Measurements Tool.

It consists on an application developed with the Kylix 3 C++ IDE that controls execution of Network Simulator (NS) [8] freeware test software packet. NS is a freely available discrete-event object-oriented network simulator, which provides a framework for building a network model, specifying data input. NS has been enhanced and used by many researchers over a couple of years. This made it a standard trusted simulator [23-25]. It is based in TCL codes and it includes traffic, agents and links libraries following the definition of described source models. A screen capture of this tool is shown in Fig. 3(b). The simulated measurements process is similar to the real one. It is automated by using a configuration input file (*Tsim.in*, that contains suitable values for the agents and links properties, obtained from real captures or theoretical studies) and a configurable network setting (*Tsim.tcl*, that generates the complete environment under simulation). The tool allows generating multiple *Tsim.in* files very easily.

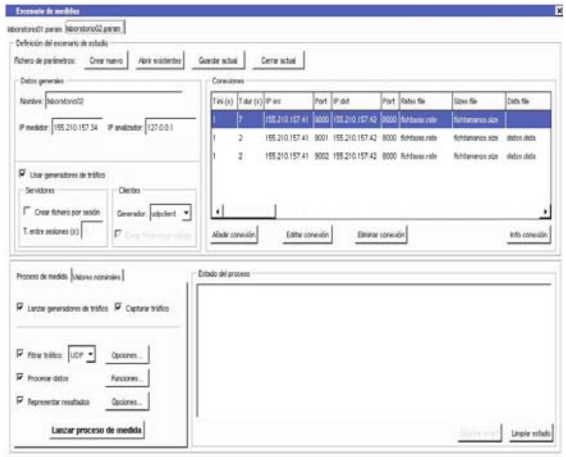
Thus, different simulations can be performed by means of an automatically created TCL script. Subsequently, simulation process is launched through following steps:

1. Event Capturing. This part of the application launches NS execution of *Tsim.tcl* file obtaining the data it needs from *Tsim.in*. The user can select one or more *.in* files, so more than one NS execution may take place. Each execution will generate certain event *timestamps* that will be written by NS software in an event output file (*Tsim.out*), one for every single *.in* file.
2. Event Filtering. The simulation tool filters the traffic obtained in *Tsim.out* file once the user has selected the flow, sending node and receiving node identifiers, obtaining the generated packet information in those nodes. It also filters the ACK information so it is not taken into consideration. The *timestamps* obtained after filtering are used to calculate some traffic values such as inter-packet time in sending and receiving ends, dropped packets ratio, etc. These data are presented in a suitable format (*Tsim.res*, which is equivalent to *Treal.res*). As in the event capturing, several *.res* files may be obtained if multiple executions have been performed.

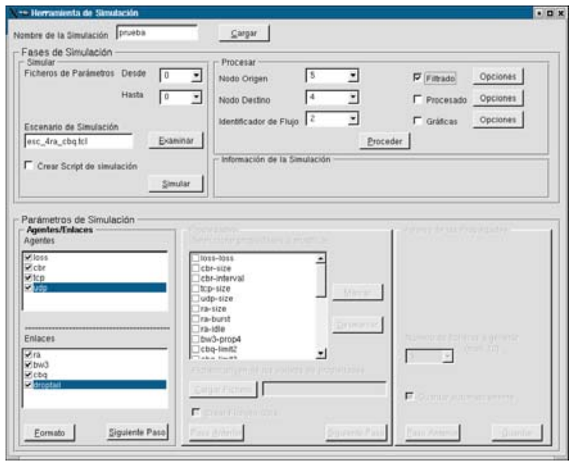
C. Results Evaluation.

Finally, the results obtained from real and simulated measurements are evaluated in order to optimize the application design. It is divided into two final steps:

3. Telematics processing, consisting of the calculation of traffic parameters that allow analysing QoS requirements in two perspectives:
   - Application QoS. It considers the variations in parameters of Telemedicine services for a fixed network resources: different *codec* types, packet sizes (s_i), inter-packet times (Δt_i), idle/busy times (d_{OFF} / d_{ON}), compression ratios, etc.
   - Network QoS. It considers the variations in parameters of network resources for a fixed application: different cell loss ratios (*CLR*), effective link capacities (bandwidths, *BW*), available BW (*aBW*) for related source traffic, and several delays (end-to-end (*CTD*), round trip (*RTD*), propagation (d_{prop}), switching and processing (p^i), queuing (q^i), service and access (d^i) and so on).
4. Graphics and statistics. The results obtained are finally represented in a useful format. This step is significant since the designed functions allow a first visual validation based on optimal thresholds traffic tendencies, bursting, etc. Thus, in a direct way results are analysed, interpreted and compared (not only between different tests of the same setting but the same test in several scenarios).



(a)



(b)

Fig 3. Test Prototype. (a) Real measurement tool (b) Simulated measurement tool.

	s_3 = 512		s_2 = 1380		s_1 = 1472	
Δt_i	CLR	BW*	CLR	BW*	CLR	BW*
42	–	30.58	43.97	sat	53.92	sat
60	–	64.03	30.38	sat	34.18	sat
76	–	74.17	11.85	sat	16.67	sat
90	–	82.53	–	5.45	1.35	sat
100	–	87.07	–	17.69	–	11.29
126	–	95.53	–	51.92	–	35.43

Table 2. CLR(%) and remaining BW, BW (Kbps) for s_i = 1380, 1472bytes; Δt_i = 90, 100ms.*

VI. Examples of Services Evaluation

The technical evaluation system has been used for evaluating of several very representative telemedicine services, based on different technological environments. Thus, a scenario that integrates RT applications (videoconference and biomedical signal transmission) with NRT services (clinical consults and off-line transfer) was evaluated over the simulated measurements tool, as a complete example of the proposed methodology. The principal aim of this test was choose the best values of RT transmission (*codec* type with suitable packet sizes and bit rates) according to the available network resources in a particular application and network scenario.

Thus, NRT service was firstly analysed to establish *aBW* thresholds and data size restrictions (around hundred of Kbps and below 1500 bytes), conditioned by TCP flow control. Besides, by fixing these values as “boundary conditions” and by calculating the optimum relation for application parameters regarding to evolution of the health network resources, a maximum delay threshold was evaluated to fulfil RT requirements.

End-to-end CTD evaluation is discussed in Table 2. Table 2, for different values of RT packet size (s_i) and of RT inter-packet time (Δt_i) corresponding to *aBW* resources (from 64 to 256Kbps), indicates CLR (in %) and remaining BW (BW*) below link capacity to avoid saturation (*sat*), for selected values. Thus, a RT transmission characterised by s_1=1472bytes and Δt_1=90ms, over a link of 128Kbps cited RT transmission, will imply a CLR=1.35% (above desirable limit of 1%), that it will be avoided by means of proposed Δt_i increase or s_i decrease. However, Table 2 shows that first option is more adequate (it allows BW*=11.29Kbps) than second one (only BW*=5.45Kbps).

Finally, optimum number of simultaneous connections (n_c) was analysed. Results showed that transmission rate values what avoid congestion situations were below 32Kbps (with *aBW* = 384Kbps and mean value of n_c =2.3).

VII. Conclusions

A methodology for technical evaluation of QoS traffic requirements in telemedicine services has been proposed. It includes service definition (both application requirements and network topologies), service modelling (from real data flow and theoretical data traces over real devices or simulated equipments conditions) and a measurements tool that allows in a direct way parameter analysis, tendency interpretation, model validation and real and simulated behaviour comparison.

Several representative telemedicine services, based on different technological environments, have been discussed in an integrated scenario including RT applications (videoconference and biomedical signal transmission) and NRT services (clinical consults and off-line transfer). Thus, the proposed work constitutes an automated and versatile methodology of technical evaluation that allows measuring QoS requirements (both application and networking ways) of telemedicine services and optimizing application design (e.g. by adjusting *codec* types, compression ratios, number of simultaneous connections, suitable packet sizes or bit rates) regarding to the network resources and fulfilling delay thresholds.

Further researches will go deeply into advanced telematics processing, best modelling of telemedicine services and to infer network tomography.

VIII. Bibliography

[1] R. Holle and G. Zahlmann, "Evaluation of telemedical services", IEEE Trans Inf Technol Biomed, vol. 3, no. 2, pp. 84-91, 1999.

[2] W.C. Hardey, "QoS Measurement and Evaluation of Telecommunications Quality of Service", Eds. John Wiley, Handcover, 230 pages. [Book Review – R. Chodoreck, IEEE Communications Magazine, 40(2):30-32, 2002].

[3] W. Jiang, K. Koguchi and H. Schulzrinne, "QoS evaluation of VoIP end-points", IEEE International Conference on Communications ICC'03, vol. 3, pp. 1917-21, 2003.

[4] M. Jain and C. Dovrolis, "End-to-end available bandwidth: measurement methodology, dynamics, and relation with TCP throughput", Proc. SIGCOMM, 2002.

[5] G. Fortino and L. Nigro, "A methodology centered on modularization of QoS constraints for the development and performance evaluation of multimedia systems", Proceedings 33rd Annual Simulation Symposium SS'00, pp.177-184, 2000.

[6] X.Fang and D. Ghosal, "Performance modelling and QoS evaluation of MAC/RLC layer in GSM/GPRS Networks", IEEE International Conference on Communications ICC'03, vol. 1, pp. 271-275, 2003.

[7] T. Yamazaki and J. Matsuda, "Adaptive QoS management for multimedia applications in heterogeneous environments: A case study with video QoS mediation", IEICE Trans. Commun., vol. E82-B, no. 11, pp. 1801–07, 1999.

[8] A. Vogel, G. Bochmann, R. Disallow, J. Geckos and B. Kerherv, "Distributed Multimedia Applications and Quality of Service – A survey", IEEE Multimedia, 1994.

[9] E. Topalis et al, "A Novel Architecture for Remote home Automation e-Services on an OSCGi Platform via High-Speed Internet Connection Ensuring QoS Support by Using RSVP Technology", IEEE Transactions on Consumer Electronics, vol. 48, no.4, pp.825-833, 2002.

[10] A. Mercado and K.J. Ray Liu, "Adaptive QoS for Wireless Multimedia Networks Using Power Control &Smart Antennas", IEEE Transactions on Vehicular Technology, vol. 51, nº 5, pp. 1223-33, 2002.

[11] X. Henocq and C. Guillemot, "Source Adaptive Error Control for Real-time Video over the Internet", International Symposium on Image/Video Communication over Fixed and Mobile Networks, ISIVC, 2000.

[12] J. Capone and I. Stavrakakis, "Delivering QoS requirements to traffic with diverse delay tolerances in a TDMA environment", IEEE/ACM Trans. Networking, vol. 7, pp. 75–87, 1999.

[13] H. Morikawa, T. Kajiya, T. Aoyama, and A. Campbell, "Distributed power control for various QoS in a CDMA wireless system", IEICE Trans. Fundamentals Electron., Com. Comput. Sci., pp.2429–36, 1997.

[14] N. Hu and P. Steenkiste, "Evaluation and Characterization of Available Bandwidth Probing Techniques", IEEE Journal on Selected Areas in Communications, vol. 21, no. 6, pp. 879-894, 2003.

[15] W.R. McDermott et al., "Optimization of Wide-Area ATM and Local-Area Ethernet/FDDI Network Configurations for High-Speed Telemedicine Communications Employing NASA's ACTS", IEEE Network, 13(4):30-38, 1999.

[16] I. Martínez, J. Salvador, J. Fernández and J. García, "Traffic requirements evaluation for a Telemedicine network", I International Congress on Computational Bioengineering, pp. 389-394, 2003.

[17] M.E. Crovella and A. Bestavros, "Self-Similarity in WWW. Evidence and Possible Causes", *IEEE/ACM Transactions on Networking*, vol. 5, no. 6, pp. 835-846, 1997.

[18] A. Reyes. "Modelado de tráfico de clientes WWW", PhD Thesis, Univ. Málaga, 2001.

[19] T.J. Kostas et al. "Real-Time voice over packet-switched networks", *IEEE Networks*, vol.12 no. 1, pp. 18-27, 1998.

[20] TCPDUMP software. http://www.tcpdump.org/

[21] RFC2330. http://www.ietf.org/rfc/rfc2330.txt.

[22] http://www.isi.edu/nsnam/ns/ns-documentation/

[23] S. Ubik and J. Klaban,"Experience with using simulations for congestion control research",CESNET Technical Report 26/2003.

[24] S. Yan, M. Faloutsos and A. Banerjea, "QoS-Aware Multicast Routing for the Internet: The Design and Evaluation of QoSMIC", IEEE/ACM Transactions on Networking, vol. 10, no. 1, 2002.

[25] A. Calveras and A. Berdonces, "QoS en redes Wireless LAN IEEE-802.11", JITEL 2003.