

Application Parameters Optimization to Guarantee QoS in e-Health Services

I. Martínez, J. García, E. Viruete, J. Fernández

Abstract— The wide development of multimedia clinical applications and the use of inter and intra-hospital communication networks require a specific analysis to increase the efficiency of e-Health services. In this paper we study the optimum combinations of the application parameters needed to fulfill the Quality of Service (QoS) thresholds according to monitored network measurements in the new healthcare services. A remote diagnosis service has been evaluated establishing good-performance areas, depending on available resources.

I. INTRODUCTION

TELECOMMUNICATIONS and advanced information technologies have increasingly been used for clinical activities and research to improve health care delivery. These technologies have undergone many investigations to evaluate their efficiency, reliability and feasibility [1], [2]. The technical evaluation firstly requires a measurement methodology in order to analyze application and network requirements and, consequently, to optimize service design according to the available resources [3], [4].

Service design is considered as definitive for the correct implementation, performance and maintenance of any service [5]. This design process permits to determine the available network resources, to characterize the volume of information to be transferred, the Type of Service (ToS) associated to the applications and the Quality of Service (QoS) requirements, and to tune and model the applications in order to obtain their optimum performance [6].

The implementation of a new e-Health service requires a technical evaluation to study its behaviour under different network conditions. In order to measure the QoS parameters and to model the multimedia service, an automated tool has been developed in previous works [7]. It includes three main components: definition and configuration (to translate the clinical requirements into technical multimedia parameters), service measurement (to capture experimental and simulated traffic), and service evaluation (including its characterization, modelling and optimization).

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This tool follows an extended idea [8]: adapting application parameters to fit network performance in terms of Packet Loss Rate (PLR), End-to-End Delay (EED) and Bandwidth (BW). This procedure has been widely developed in multimedia scenarios over best-effort networks, such as Internet, but a specific analysis over telemedicine environments, like the study presented in this paper, would permit to improve the quality of e-Health communications [9].

In this paper, a technical evaluation of traffic parameters to guarantee QoS requirements in a remote diagnostic service is proposed. The main characteristics of the scenario under study, including the most suitable applications and network technologies, are given in Section II. The results obtained under different network conditions are analyzed in Section III in order to characterize, model and optimize the telemedicine service.

II. MATERIALS AND METHODS

This work presents an evaluation setting focused on a hospital scenario associated to the inter-disciplinary communications among health professionals of different specialties (even belonging to different hospitals) to do cooperative work, share medical applications, remote diagnosis, etc. These communications are usually supported by well-known wired technologies such as Frame Relay (FR) Asynchronous Transfer Mode (ATM) or Ethernet. These network resources are usually shared among many multimedia applications. They include different ToS and need specific QoS requirements. The ToS are usually grouped into two main categories: Transmission Control Protocol (TCP)-based services, for reliable data transfer (e.g. biomedical information transfers and medical files retrieval); and User Datagram Protocol (UDP)-based services, for interactive applications (e.g. audio and video-conference, and *on-line* signals transmission).

Therefore, an evaluation of the main TCP/UDP *cross-traffic* situations becomes necessary in order to manage the intermediate *buffers* (flow control methods, Active Queue Management (AQM) algorithms, queue size (Q), etc.), and to select the parameter combinations that improve service performance.

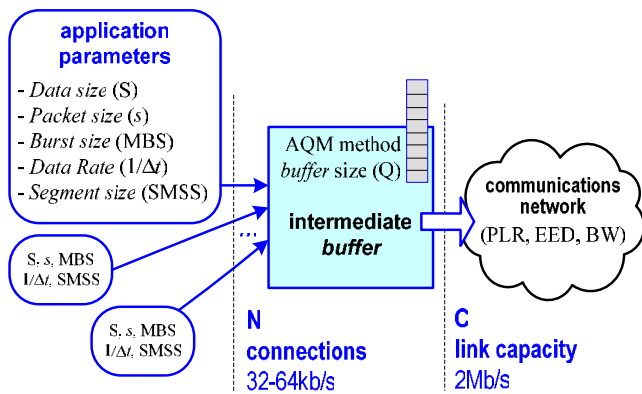


Fig. 1. Generic evaluation scheme, including application parameters, network conditions and the main QoS measurements in e-Health services.

This evaluation scheme (Fig. 1) is completed with the specific parameters, transfer rates and ToS models used (Table I) to define the complete scenario that distinguishes four main use cases (UCs):

- **UC1.** The health professional sends biomedical data to the hospital in Store-and-Forward (SF) mode. This SF.Data service includes medical tests and patient-related information.
- **UC2.** Including UC1, the health professional also queries the hospital database to manage the Electronic Patient Report (EPR) through a real-time (RT) web connection (RT.EPR).
- **UC3.** Including UC2, the health professional establishes a RT multimedia conference (including audio and video) with other specialist to support the diagnosis (RT.Media).
- **UC4.** The health professional often requires acquiring and sending specific biomedical signals in real-time (electrocardiography (ECG), ecography (ECO), etc.) in order to complete the patient diagnosis (RT.Bio).

UC1 and UC2 require the study of the main TCP parameters: protocol version, congestion window size (*cwnd*), Slow Start (SS) algorithm threshold (*ssthresh*), Sender Maximum Segment Size (SMSS), or Transmission time (T). On the other hand, UC3 and UC4 require the study of the main UDP parameters: data size (S), packet size (s), Maximum Burst Size (MBS) and data transmission rate ($1/\Delta t$).

All these use cases are included in a tele-diagnosis service (Fig. 2) which can be characterized, modelled and optimized, in order to obtain the good-performance QoS thresholds.

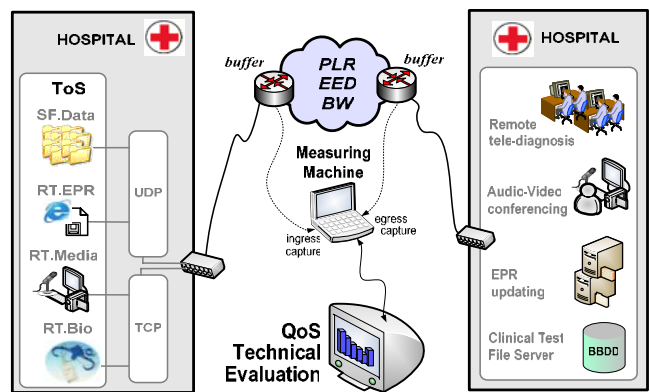


Fig. 2. Scenario to evaluate a telemedicine service based on tele-diagnosis between hospital centres, including medical tests and patient-related information transfer, EPR updating, audio and video conference, and biomedical signals transmission.

TABLE I. PARAMETERS UNDER STUDY AND TOs MODELS

Use Case	e-Health service	ToS	rates	parameters	
UC4	UC1	patient info	SF.	TCP parameters protocol version, <i>cwnd</i> - <i>ssthresh</i> SS algorithm, packet size (SMSS) trx. time (T)	
		medical transfer	Data		
	UC2	EPR management	RT.		8kb/s
		database query	EPR		10kb/s
UC3	audioconference	RT.	16kb/s	UDP parameters data size (S), packet size (s), burst size (MBS), trx. rate($1/\Delta t$) user number (N)	
	videoconference	Media	40kb/s		
		biomedical signals	RT.	4kb/s	
		RT transmission	Bio	24kb/s	

III. RESULTS AND DISCUSSION

A. Evaluation of TCP services (UC1 and UC2)

This evaluation implies selecting the optimum value among a large range of specific values for the parameters involved. Several works published [10] have concluded that the combination of Vegas as TCP version and Class Based Queuing (CBQ) as AQM method, contributes to improve service behaviour. In previous studies of this work [11], only focused in UC1, we obtained significant results to select the best values of *cwnd* and *ssthresh*: distant and high values imply low performance, and close and low values activate the SS algorithm and induce continuous oscillations in link utilization. That is, they directly depend on the selected SMSS and the scenario under study.

Using these previous results, UC1 and UC2 were evaluated analyzing the study parameters shown in Table I with the QoS network conditions recommended by the International Telecommunications Union (ITU) ($PLR < 0.20$ and $EED < 180s$) [12]. This evaluation includes specific test sizes (S) for the main medical practices: computing radiography, electrocardiography, ecography, etc.

First, the results obtained show that T decreases when SMSS rises, and in a nonlinear way with the loss variability: T practically remains constant with a low PLR level ($PLR < 0.05$), but T notably increases (3:1) when PLR rises (see Fig. 3 with $S=384MB$ related to an ECO basic study). Since large packets fit the link capacity better, T decreases with higher SMSS values. However, in those cases, the retransmission percentage increases, due to the fact that the SS algorithm depends on the number of packets received (regardless of their size) and fills the intermediate *buffer*.

Then, the EED analysis (Fig. 4) keeps this compromise: it is almost constant without PLR (independently of SMSS) but rises when SMSS does (due to more SS algorithm reactivations with lower SMSS values that empty the *buffer* and reduce EED).

Finally, Fig. 5 shows the available BW (ABW) depending on PLR level. With low SMSS values, BW decreases down to $ABW=1.4Mb/s$ in the worst case (with $SMSS=512B$ and $PLR=0.20$).

In summary, all these tendencies recommend the use of low SMSS values in order to avoid using specific priority allocation methods and allow sharing network resources with UDP services, as it is evaluated in the next section.

B. Evaluation of UDP services (UC3 and UC4)

The UC3 and UC4 were evaluated again with the specific parameters shown in Table I and the network thresholds recommended for RT services ($EED < 200ms$, $PLR < 0.15$) [12]. The evaluation includes specific RT data sizes $S_i = \{4k, 2k, 1k, 500 (B)\}$, a variable number (N) of simultaneous user connections (with user rate $r_i = \{64, 32 (kb/s)\}$), a *buffer* size range of $Q \in [5, 15]$, and two groups of packet sizes: large packets, $s_{Hi} = \{1472, 1380 (B)\}$, and small packets, $s_{Li} = \{512, 240 (B)\}$.

First, considering a user rate $r_1 = 64kb/s$, the PLR analysis shows for $s_{H1} = 1472B$ (upper) and $s_{L2} = 240B$ (down), the situations that fulfill the PLR threshold (white cells in Table II). These combinations are:

- Large packets: $Q \geq 10$ (with $N=4$) and $Q \geq 6$ (with $N=2$).
- Small packets: $Q \geq 15$ (with $N=4$) and $Q \geq 15$ (with $N=2$).

When priority allocation (based on the CBQ algorithm) is included in the study, a decrease in the PLR was observed. Thus, the range of Q that guarantees QoS is modified in the following way:

- Large packets: $Q \geq 8$ (with $N=4$) and $Q \geq 5$ (with $N=2$).
- Small packets: $Q \geq 12$ (with $N=4$) and $Q \geq 12$ (with $N=2$).

The EED analysis includes the variation of link capacity by evaluating when QoS thresholds are guaranteed (Fig. 6 shows a representative case with $S_1=4kB$ and $N=4$):

- Large packets: if $C \geq 128kb/s$, $Q \leq 5$; if $C \geq 192kb/s$, $Q \leq 7$;
if $C \geq 256kb/s$, $Q \leq 9$; if $C \geq 384kb/s$, $Q \leq 12$.
- Small packets: if $C \geq 128kb/s$, $Q \leq 8$; if $C \geq 192kb/s$, $Q \leq 10$;
if $C \geq 256kb/s$, $Q \leq 11$; if $C \geq 384kb/s$, $Q \leq 15$.

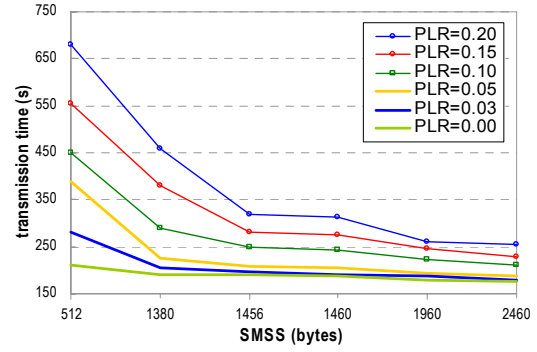


Fig. 3. Evolution of T as a function of SMSS, with variable PLR.

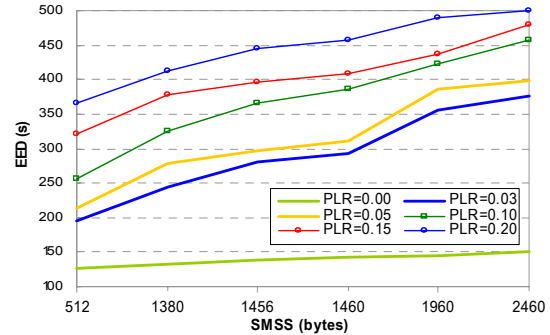


Fig. 4. Evolution of EED as a function of SMSS, with variable PLR.

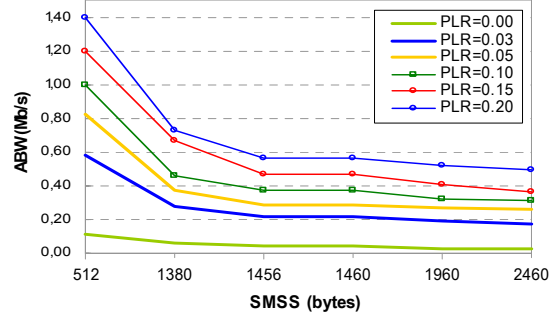


Fig. 5. Evolution of ABW as a function of SMSS, with variable PLR.

TABLE II. EVOLUTION OF PLR AS A FUNCTION OF Q WITH DIFFERENT $\{S, N\}$ COMBINATIONS AND TWO GROUPS OF PACKET SIZES:

(A) LARGE PACKETS						
s_{H1}	$S_1=4kB$			$S_2=2kB$		
	$N=1$	$N=2$	$N=4$	$N=1$	$N=2$	$N=4$
$Q=15$	0.023	0.028	0.096	0.009	0.027	0.037
$Q=12$	0.032	0.043	0.126	0.012	0.034	0.044
$Q=10$	0.044	0.051	0.144	0.023	0.046	0.049
$Q=8$	0.055	0.072	0.197	0.027	0.052	0.056
$Q=7$	0.066	0.108	0.214	0.031	0.057	0.221
$Q=6$	0.073	0.144	0.241	0.039	0.064	0.243
$Q=5$	0.081	0.222	0.274	0.042	0.223	0.263

(B) SMALL PACKETS									
s_{L2}	$S_2=2kB$			$S_3=1kB$			$S_4=500B$		
	$N=1$	$N=2$	$N=4$	$N=1$	$N=2$	$N=4$	$N=1$	$N=2$	$N=4$
$Q=15$	0.120	0.150	0.15	0.06	0.12	0.14	0.02	0.08	0.13
$Q=12$	0.140	0.190	0.22	0.09	0.15	0.15	0.02	0.12	0.15
$Q=10$	0.150	0.230	0.25	0.13	0.21	0.22	0.03	0.15	0.19
$Q=8$	0.170	0.240	0.29	0.15	0.22	0.25	0.04	0.17	0.22
$Q=7$	0.210	0.260	0.31	0.18	0.22	0.26	0.04	0.18	0.24
$Q=6$	0.230	0.280	0.35	0.21	0.24	0.29	0.05	0.21	0.26
$Q=5$	0.260	0.300	0.37	0.24	0.26	0.32	0.18	0.24	0.28

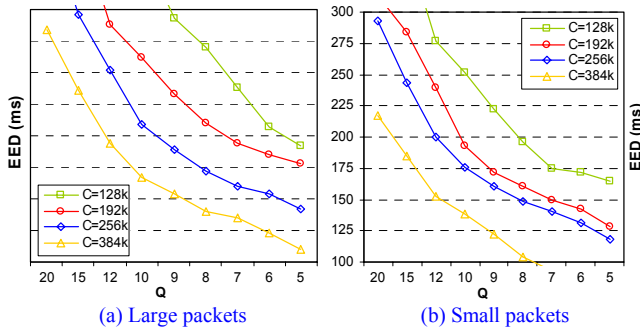


Fig. 6. Variation of EED as a function of Q (with $r_1=64\text{kb/s}$ and variable C), focusing on the most critical areas closer to the QoS thresholds around the recommended $EED=200\text{ms}$.

This EED analysis is completed with $r_2=32\text{kb/s}$, for the same S_i variation range (modifying, therefore, the ratio $S_i/\Delta t_i/MBS_i=\{4\text{kB}/1\text{s}/16, 2\text{kB}/500\text{ms}/8, 1\text{kB}/250\text{ms}/4, \text{etc.}\}$). Fig. 7 shows a representative example with $S_1=4\text{kB}$ and $s_{H1}=1472\text{B}$ (the rest of the tests show similar trends). The new EED ($r_2=32\text{kb/s}$, $N=8$) is lower than the previous ($r_1=64\text{kb/s}$, $N=4$), even with the same information amount. Moreover, it changes the QoS thresholds: $C=192\text{kb/s}$ is now useful with $Q\leq 9$ (previously $Q\leq 7$), and $C=384\text{kb/s}$ is useful with $12\leq Q\leq 15$ (previously only $Q\leq 12$). Thus, low burst sizes fulfill better the QoS requirements.

In summary, this evaluation allows establishing several good-performance areas, depending on available resources, and adjusting these parameters, according to monitored network measurements (EED, PLR, BW), to guarantee QoS.

Finally, Fig. 8 shows an example of how the adaptive selection of the *buffer* size would be, depending on priority allocation, packet size, connected users and link capacity. The *buffer* is initially sized with $Q=15, 10$, or 6 depending on $N=4/N=2$ with small packets (step {1a}), $N=4$ with large packets (step {1b}), or $N=2$ with large packets, respectively, considering $C=256\text{kb/s}$. If the link capacity decreases to $C=192\text{kb/s}$, the *buffer* size can be reconfigured to $Q=12, 8$, or 5 , (steps {2a}, {2b} or {2c}, respectively).

IV. CONCLUSIONS

In this paper, the specific application parameters of the e-Health services considered in the study have been evaluated in order to adapt it to available resources. The proposed evaluation has been focused on a tele-diagnosis service in a hospital scenario that combines TCP and UDP-based services. The results obtained show the best performance range with small TCP packets and *buffer* sizes lower than 15 packets for the UDP services. These values have been selected according to monitored network measurements and are recommended in further designs of e-Health services in order to guarantee the specific QoS requirements.

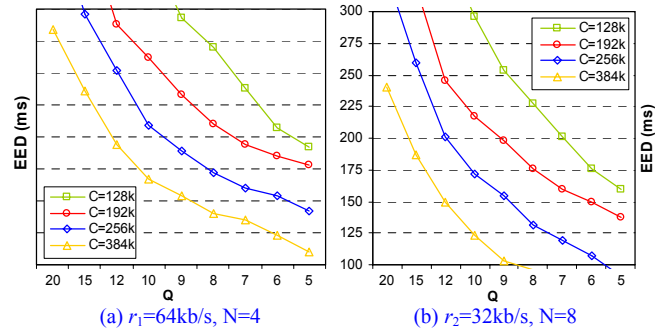


Fig. 7. EED values using different user transmission rates (with large packets and variable C), focusing on the most critical areas closer to the QoS thresholds around the recommended $EED=200\text{ms}$.

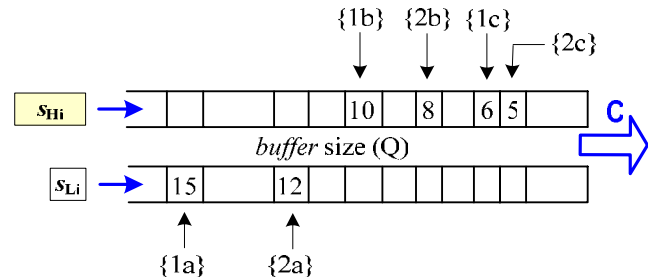


Fig. 8. Adjustment scheme to select Q depending on available resources.

REFERENCES

- [1] M.F.M.T. Du Molin et al., "The reliability of diagnosis using store-and-forward teledermatology", *Journal of Telemedicine and Telecare*, vol. 9, no.5, pp. 249-252, 2003.
- [2] C. Sicotte, P. Lehoux, J. Fortier-Blanc and Y. Leblanc, "Feasibility and outcome evaluation of a telemedicine application in speech-language pathology", *Journal of Telemedicine and Telecare*, vol. 9, no.5, pp. 253-258, 2003.
- [3] A. Schneider, D. Wilhelm, U. Bohn, A. Wichert and H. Feussner, "An evaluation of a surgical telepresence system for an intrahospital local area network", *Journal of Telemedicine and Telecare*, vol. 11, no.8, pp. 408-413, 2005.
- [4] P. Taylor, "Evaluating telemedicine systems and services", *Journal of Telemedicine and Telecare*, vol. 11, no.4, pp. 167-177, 2005.
- [5] J. Major, "Telemedicine room design", *Journal of Telemedicine and Telecare*, vol. 11, no.1, pp. 10-14, 2005.
- [6] 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.
- [7] I. Martínez and J. García, "SM3-Quality of Service evaluation tool for Telemedicine-Based New Healthcare Services", *International Congress on Computational Bioengineering ICCB*, pp.1163-73, 2005.
- [8] 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].
- [9] E. A. Viruete, J. Ruiz, J. Fernández, C. Peña, "Performance of a 3G-Based Mobile Telemedicine System", *Proceedings of the IEEE Consumer Communications and Networking Conference (CCNC'06)*, vol. 2, pp. 1023-1027, 2006.
- [10] Stephen H. Low, "A duality model of TCP and queue management Algorithms", *IEEE/ACM Trans. Net.*, vol. 11, no. 4, Aug. 2003.
- [11] I.Martínez, J.Salvador, J.Fernández, J.García, "Traffic requirements evaluation for a Telemedicine network", *I International Congress on Computational Bioengineering ICCB'03*, pp. 389-394, 2003.
- [12] D. Wright, "The ITU's Report on Telemedicine and Developing Countries", *Journal of Telemedicine and Telecare*, 4(1):75-79, 1998. [Spanish version. *International Telemedicine*, 7-8, 1998].