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TRAFFIC REQUIREMENTS EVALUATION FOR A TELEMEDICINE NETWORK

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ABSTRACT

The theoretic study of inter and intra-hospital communications traffic constitutes the first step in Telemedicine network design. Real Time (RT) applications requirements and Quality of Service (QoS) parameter analysis are needed during the evaluation of a complete system. In this paper we present a test methodology definition implemented by an automated process that includes the study network setting, the traffic characteristics obtaining and the network equipment modelling, considering a simulation environment based on *Network Simulator* (NS) software, and a real measure process, using *tcpdump* packet capture library. Results obtained establish bandwidth and data size restrictions (around hundred of Kbps and below 1500 bytes) and an optimum relation for the application model with regard to the Telemedicine network resources. The proposed method can be used to adjust traffic parameters (e.g. transferred clinical file sizes or electrocardiograph (ECG) signal transmission rates), to guarantee desirable QoS requirements.

1. INTRODUCTION

Network design is considered by diverse authors as definitive for its correct network implementation, performance and maintenance. This design process follows a logical order: first, to determine the available bandwidth in the network, then to analyse the type, volume and QoS requirements of information to be transferred, and finally to design different applications that the network is going to support and decide how to do its management. This introduces a fundamental question that theoretical study must answer: "how much bandwidth is enough?" [1].

But even more important than theoretical design is the evaluation of the behaviour of networks already implemented. Multimedia technology is going to demand too much bandwidth from network technologies and infrastructures, and its potential impact in corporative networks (e.g. hospital communications) has not been adequately evaluated [2]. And, moreover, a set of some minimum requirements need to be fulfilled due to the variable nature of Telemedicine traffic (image, audio, video). In order to establish these requirements and to evaluate them later, a certain quality of service of data generated by the different applications should be considered [3-4].

RT applications using Local Area Network (LAN) interconnection technologies is a common solution to integrate inter and intra-hospital communications. These RT transmissions imply quantitative and qualitative characteristics to guarantee suitable QoS requirements [5]. Each one of this kind of traffic applications implies different QoS parameters, and therefore requires a particular assignment bandwidth and delay analysis.

In this article, an evaluation methodology for QoS traffic requirement in Telemedicine networks is proposed. Section 2 defines the test methodology that includes study setting, traffic and network model, and complete system analysis. An overview of the Telemedicine network design based on different communication services that are provided is given in section 3. Section 4 defines simulated implementation of these traffic models based in CBR and VBR-rt source classes. In Section 5 the network environments used to create both simulation and real measure settings, are presented. Section 6 describes the system analysis automated process an selected QoS parameters. Results obtained and tests evaluation are discussed in section 7.

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2. TEST METHODOLOGY DEFINITION

There are some alternatives to develop a network traffic evaluation, based on theoretical or simulated studies. In this project, as it is shown in Fig. 1, a test methodology definition is suggested, divided into the following steps:

- ***Study system setting***. It includes the selection of:
  - Inter-hospital connection topology, corresponding to WAN environment.
  - Intra-hospital connection topology, corresponding to LAN environment.
  - Telemedicine applications to use.
- ***Application generated traffic obtaining***. It will be used in simulation test by means of data trace obtaining from:
  - Real traffic captures.
  - Traffic model, found from previous multiple data trace studies or theoretical standard studies.
- ***Network equipment modelling***. It also will be used in simulation step, consisting of:
  - Preliminary theoretical information, by characterisation of network equipment conditions (loss rate, congestion situations, buffer limitations, flow control methods, and so on).
  - Subsequent experimental information, by capturing traffic in a laboratory equipment real setting.
- ***Simulation system***. It implies the use of the application data trace and the network model to obtain results that subsequently will be used to optimise the network scheduling, equipment operation, and so on.

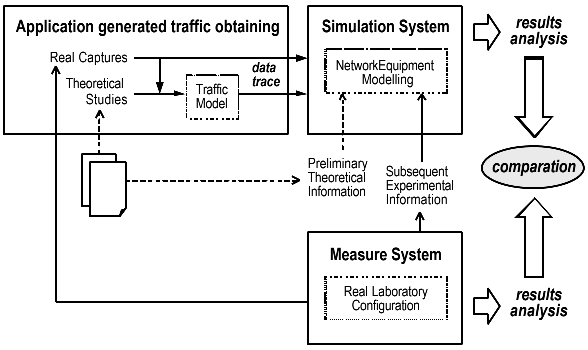

Fig.1. Test methodology basic scheme.

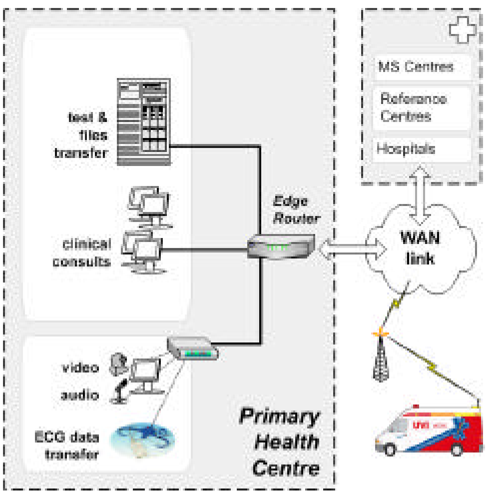

Fig.2. Hospital network model.

3. STUDY SYSTEM SETTING

A study regarding the application of telematics technologies to the sanitary environment of the Community of Aragón is presented in [6]. It develops a theoretical analysis of inter and intra-hospital desirable bandwidth to guarantee communications among centres.

Wide Area Network (WAN) inter-hospital design is based on the connection necessities of geographically dispersed regions. Frame Relay (FR) technologies, for rural zones with 2 Mbps maximum link-capability, have been considered.

LAN access design is based on the health system hierarchical structure [7]. In Fig. 2 an scheme of Primary Health (PH) centre model is shown that includes different equipment connected to WAN inter-communication edge router. In an external way, traffic from other health network nodes (as hospitals, reference centres or Medical Specialities (MS) centres) or mobile units can be considered in the global system.

Moreover, various kinds of communications during medical activities have been considered: medical test and administrative file transfer (from clinical data exchange between centres or speciality sections), biomedical signals transmission (e.g. ECG, blood pressure or oxygen saturation), patient reports and clinical routine consults (that occur during accesses to databases, queries to medical report warehouse, and so on), and multimedia applications (e.g. an inter-hospital videoconference including audio and video).

Thus, the complete study setting is shown in Fig. 3. It includes two services types: Constant Bit Rate (CBR) services for non real time applications (including continuous traffic, corresponding to data transfer, and burst traffic, corresponding to clinical consults), and Variable Bit Rate – real time (VBR-rt) services for real time applications, corresponding to a multimedia connection. Therefore, the setting allows to perform QoS evaluation in congestion situations (over a restricted inter-communication link defined between access and network nodes) in order to optimise traffic parameters for Telemedicine applications according to available network resources.

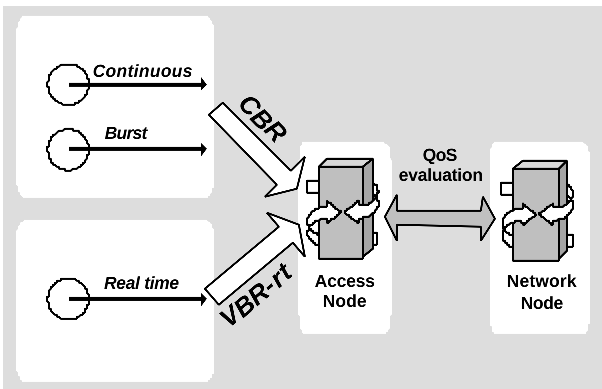

Fig.3. Study setting.

4. APLICATION GENERATED TRAFFIC OBTAINING

Simulation traffic models for this health network have been designed by using *Network Simulator* (NS) [8] freeware test software packet, based on the definition of the previously described source classes (CBR, VBR-rt). A capture of simulation environment is shown in Fig. 4.

CBR traffic service 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 Cell Rate (PCR). It defines *elastic* applications, based on an ASAP (as-soon-as-possible) model. Two traffic generators following "*best-effort*" service class have been designed:

- *FTP agent*, which generates continuous rate traffic, modelled by two constant parameters: the packet size and the source rate. The variation ranges used for these parameters come from Maximum Segment Size (MSS) considered from clinical test obtained in [6].
- *WEB agent*, which generates burst traffic, modelled by two variable parameters: packet size, following a Pareto distribution based on "*heavy tail*" assumption [9], and inter-cells time, following an exponential distribution based on Markov models [10].

VBR-rt traffic service is basically defined over User Datagram Protocol (UDP), to provide with a connection characterised by statistical gain and small nonzero random loss ratio, parameterised by a Sustained Cell Rate (SCR). It defines *non-elastic* applications (that are “no waiting data arrival” later than a threshold time). A real time traffic generator, *RT agent*, following multimedia service class has been designed to generate variable rate traffic characterised by a burst exponential source. It is modelled by two constant parameters [11]: packet size (UDPsize) and inter-cell time (UDPrate), both conditioned by Burst Tolerance (BT), that determines the Maximum Burst Size (MBS) or number of end-to-end cells that can be sent at a PCR, see Ec. 1:

$$BT = (MBS - 1) \cdot \left(\frac{1}{SCR} - \frac{1}{PCR}\right) \quad (cells) \qquad (1)$$

In the variation range of UDPsize values as 240bytes (for MBS=24 in G.729 audio transmission standard), 1380 bytes (for Virtual Private Networks) or 1472 bytes (for Ethernet, 1500 bytes minus 18 bytes of overhead) were considered. The UDPrate was selected ranging from 42ms to 126ms (e.g. for videoconference [12]).

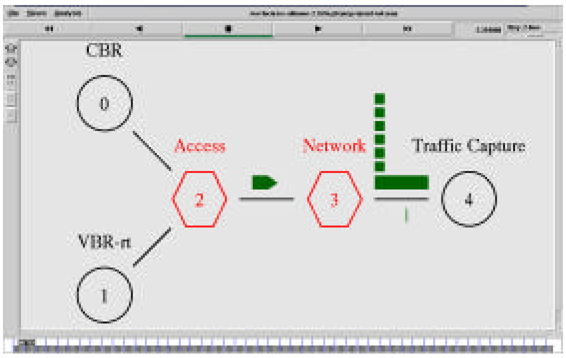

Fig 4. Simulation environment over NS.

5. NETWORK EQUIPMENT MODELLING

The characterisation of network equipment conditions has been defined in two different ways:

- ***Preliminary theoretical information***, based on a NS software configuration that includes restrictive conditions in congestion situations for the FR inter-connection link [12] to consider scheduling algorithms (based on Stochastic Fairness Queuing, SFQ), propagation times (below 150ms for WAN links), loss probability models (below 1%), etc.

- ***Subsequent experimental information***, based on a FR technology to consider real behaviour in a telematics laboratory environment. It includes a generator and a receiver, both connected through the system under test, and a traffic analyser to obtain suitable measurement parameters, as it is presented in Fig. 5. All computers mount identical network cards, using Linux as operating system and *tcpdump* [13] software running with *libcap* packet capture library (which uses Berkeley Packet Filter (BPF) system to capture traffic cells from both network ends). The measure process, following RFC 2330 [14] of IPPM (*IP Performance Metric*) workgroup, is divided into four principal steps, shown in Fig. 5:
  - *Packet Capture*. It defines the capture process that is limited by several factors: network card interruptions, process changes, bytes transferred to user space, and system load from other processes.
  - *Timestamp*: It defines the temporal stamp that *libcap* applies to every packet, following ASAP assignation from *kernel*. Clock reference is the same for two *tcpdump* processes launched in each test point; so, end-to-end synchronisation is guaranteed.
  - *Packet Filtering*: It defines the traffic pre-selection of captured packets, by means of several *tcpdump* options, to reduce output data amount. This process occurs after than *timestamp* event to guarantee accuracy based on ASAP assignation.
  - *Result Transfer*: It defines the calculation process of interest differential parameters (e.g. delay) that must be transferred to compute in a common machine. It is desirable to use analyser as store&process machine in order to reduce the network total load.

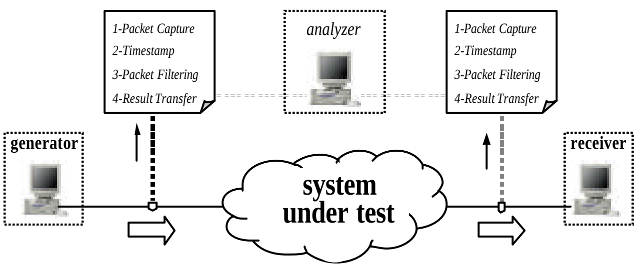

Fig 5. Measure process scheme.

6. SIMULATION SYSTEM

The final step of the test methodology implies to analyse QoS parameters for evaluating the requirements of each defined class service. Global system analysis has been automated by executing a C++ language program, and by using a configuration input file (which includes both traffic parameters and network characteristics model) that allows to be used both over the NS environment and over the telematics laboratory.

Thus, first step of this automated process consists on *traffic capture*. It can be obtained from NS environment (by an executable script that selects temporal events to create a data output file), or from real measure tests (by *tcpdump* software, described in Section 5, that selects timestamps to create a data output file, with the same format than the corresponding NS file). Following steps, which are common for both environments, include *data processing* (calculation of traffic differential values such as inter-cell time, delay, loss ratio, etc.) and *results analysis* (graphics and statistics). Interpretation and comparative of these results allow to characterise traffic evolution and network behaviour to evaluate the system.

The following QoS parameters have been considered in the automated test process:

- *Cell Loss Ratio* (CLR): the percentage of cells that are lost in the system and are not delivered to end.
- *Bandwidth* (BW): is referred to as the traffic used for related class source traffic regarding to link capacity. Usually is also defined *throughput* as bandwidth occupation or data flow sharing.
- *Cell Transfer Delay* (CTD): the end-to-end delay experienced by a cell, which includes propagation, queuing, and service times at intermediate switches. Also a measure of CTD variance is referred to as *jitter*; it is a basic value in RT traffic since higher variation implies larger buffering.

7. RESULTS AND DISCUSSION

CBR service has been firstly analysed to evaluate *throughput* and to establish available BW and data size restrictions (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 has been evaluated to fulfill RT requirements.

7.1 CBR Service Behaviour

CBR analysis is based on TCP congestion control that usually is defined by means of the Slow Start (SS) algorithm [15] to consult and detect the network state and to shape the data flow. It is parameterised by a congestion window (*cwnd*), and a threshold value according to the available bandwidth (*ssthresh*, which is initialised by default to 65535 bytes, the maximum window size). They operates by observing that the rate at which new packets should be injected into the network is the rate at which the acknowledgments (ACK) are returned by the other end.

This provides an multiplicative growth (because the receiver typically sends one ACK for every two segments that it receives), that it becomes linear when threshold value is smaller than congestion window. This behaviour has been evaluated over NS (see Fig. 6), corresponding to the following implementation scheme:

```
1. Send one packet (at the beginning, or after a timeout)
2. For each received ACK, send two packets.
3. Every time a new ACK is received:
   if (cwnd < ssthresh) then
         cwnd+ = 1;          /*multiplicative increase*/
   else  cwnd+ = 1/cwnd;  /*linear increase*/
4. If a timeout occurs (a packet has been dropped)
   ssthresh = cwnd/2;  cwnd = 1;
```

The SS algorithm effect has been evaluated in a connection with loss (CLR≠0), to obtain the MSS dependence, as it is shown in Fig.7. The CTD is smaller when CLR increases, due to each SS activation implies a continuous sending packet brake corresponding to a transitory buffer liberation that decreases the queuing waiting time and, therefore, the mean CTD. If buffer size variation is considered, an optimum queue threshold value can be calculated to minimise packet loss due to link congestion. Moreover, due to the CTD increases when MSS does, selection of smaller MSS is desirable.

Bandwidth occupation in congestion situations, by considering CBR services with SFQ scheduling assignation was evaluated. Figure 8 shows that the BW increases when MSS does but in a non-linear way. In fact, due to the use of the SS flow control algorithm, the CLR influence is more determinant than MSS is for packets smaller than 1500 bytes. Above this value, the 2Mbps FR link is practically full, independently of the CLR since a larger data transmission exceeds the total occupation. Below it, the CLR effect is significant due to each lost cell activates the SS algorithm, which reduces the sending rate (dramatically for smaller sizes and larger CLRs), and provides an available BW around hundred of Kbps. Moreover, this SS effect is minimised by queuing, (see trace in Fig. 8 corresponding to CLR=0% buffer) that guarantees cells existence in buffer and, therefore, always-occupied link.

7.2 VBR-rt Service Behaviour

Following the CBR results that establish restrictions for available BW and MSS thresholds (below 300Kbps and 1500bytes, discussed in Section 7.1, see Fig. 8), and under the assumption of adjusting to the maximum permitted delay (below 150ms), VBR-rt service has been evaluated. Thus, the mean BW evolution when modifying UDPsize and UDPrate (numerically indicated in right side), are presented in Fig. 9. Now, due to non existence of a flow control method, results show linear and directly proportional tendency (corresponding to UDP theoretical behaviour). Thus, a link with loss influences the BW in a constant decreasing factor, but it is not relevant in terms of CTD, due to non-packet retransmission.

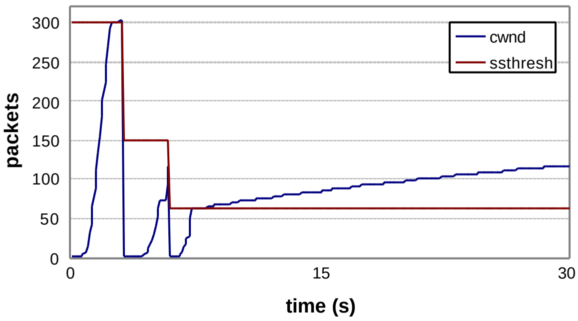

Fig. 6. SS algorithm working scheme.

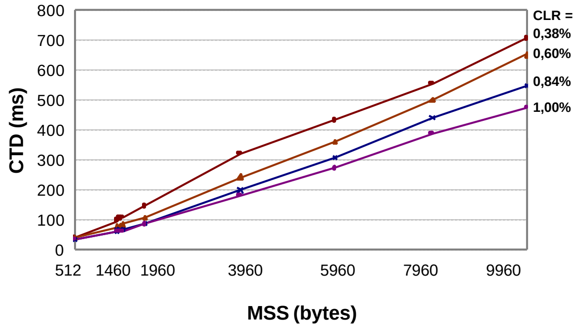

Fig.7. CTD variation in a connection with loss.

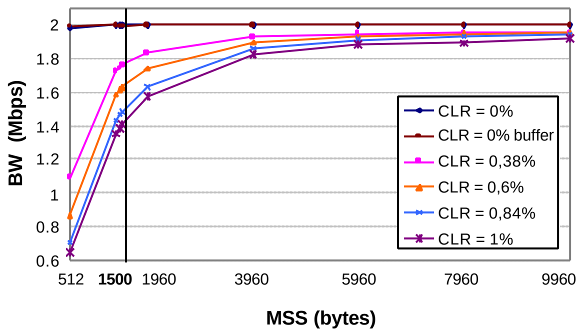

Fig.8. Mean BW variation with MSS.

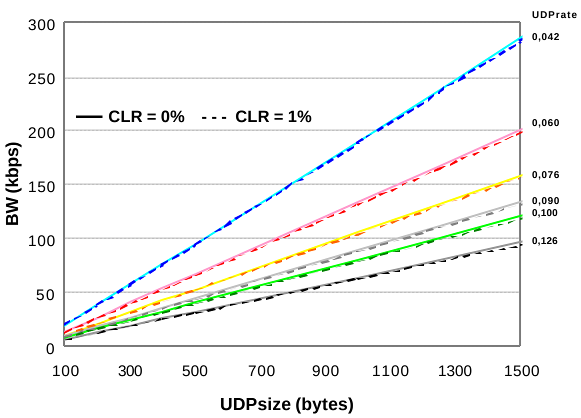

Fig.9. Throughput with regard to UDPsize and UDPrate.

Finally, end-to-end CTD evaluation is discussed in Fig. 10. Traffic results have been obtained considering FR environments described in section 5. Each graphic, for different values of UDPsize, represents CTD variation for selected values of UDPrate corresponding to available BW resources (from 64 to 256 Kbps). For example, a RT transmission characterised by UDPsize=1472bytes and UDPrate=90ms, over a link of 128Kbps, see Fig.10(f), will not fulfill a delay threshold of 150ms; it will need to increase UDPrate to 100ms, or to decrease UDPsize to 1380bytes, see Fig.10(e).

But all possible simulated values are not desirable to be selected. Table 1 indicates CLR (in %) and remaining BW (BW*) below link capacity to avoid saturation (*sat*), for the most restrictive situations (128Kbps and 64Kbps) and selected values of UDPrate and UDPsize. Following previous example, cited RT transmission over a link of 128Kbps, will imply a CLR=1.35% (above desirable limit of 1%), that it will be avoided by means of proposed UDPrate increase or UDPsize decrease. However, Table 1(a) shows that first option is more adequate (it allows BW*=11.29Kbps) than second one (only BW*=5.45Kbps).

In summary, the proposed evaluation in this work constitutes a first approach to the health networking problem by means of a simplified model both in traffic parameters (sizes and rates) and network structure (bandwidth and delay). Further research including more detailed models is needed.

UDP	***size = 512***		***size = 1200***		***size = 1380***		***size = 1472***	
rate	**CLR**	**BW***	**CLR**	**BW***	**CLR**	**BW***	**CLR**	**BW***
42	–	30.58	43.97	*sat*	43.97	*sat*	53.92	*sat*
60	–	64.03	19.96	*sat*	30.38	*sat*	34.18	*sat*
76	–	74.17	–	1.84	11.85	*sat*	16.67	*sat*
90	–	82.53	–	21.44	–	5.45	1.35	*sat*
100	–	87.07	–	32.08	–	17.69	–	11.29
126	–	95.53	–	64.08	–	51.92	–	35.43

(a)

UDP	***size = 512***		***size = 1200***		***size = 1380***		***size = 1472***	
rate	**CLR**	**BW***	**CLR**	**BW***	**CLR**	**BW***	**CLR**	**BW***
42	34.33	*sat*	71.96	*sat*	75.61	*sat*	76.94	*sat*
60	6.20	*sat*	59.95	*sat*	65.16	*sat*	67.06	*sat*
76	–	10.17	49.27	*sat*	55.86	*sat*	58.27	*sat*
90	–	18.53	39.93	*sat*	47.74	*sat*	50.60	*sat*
100	–	23.07	33.27	*sat*	41.95	*sat*	45.12	*sat*
126	–	31.53	15.98	*sat*	26.91	*sat*	30.91	*sat*

(b)

Table 1. CLR and BW remaining (BW) for links o f : (a) 128Kbps. (b) 64Kbps.*

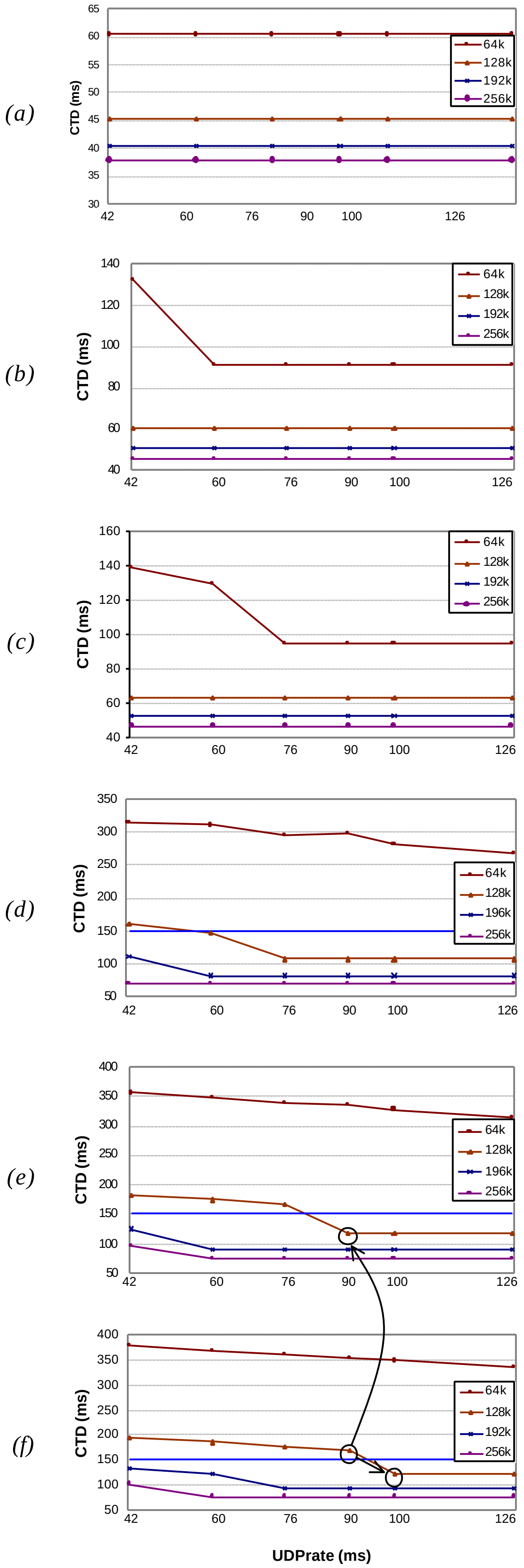

Fig.10. CTD(ms) vs UDPrate(ms) for UDPsize(bytes) of (a)240. (b)480. (c)512. (d)1200. (e)1380. (f)1472.

8. CONCLUSIONS

An evaluation methodology for QoS traffic requirements in a Telemedicine network has been proposed by defining a study setting, by modelling traffic from medical activities as CBR and VBR-rt services, by developing both simulated and real measure environments, and by analysing a complete system following an automated test process. From the CBR study, available BW and data size restrictions (around hundred of Kbps and below 1500 bytes) have been obtained. And from the VBR-rt analysis, an optimum relation for the application model with regard to the Telemedicine network resources and fulfilling delay thresholds has been evaluated. The proposed method can be used to adjust traffic parameters (e.g. transferred clinical file size or ECG transmission rate), to guarantee desirable QoS requirements.

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