Videoconference Transport in the IP Core Network of a 3G Mobile System using DiffServ

Ernesto Quiroz, Jesús Leal and Rogelio Rodríguez

Instituto Politécnico Nacional- Centro de Investigación y Desarrollo de Tecnología Digital
Av. del Parque No. 1310, Mesa de Otay, 22510 Tijuana, Baja California, México.
Phone:(664)623-1344, Fax:(664)623-1388

{eequiroz, jleal, rrodriguez}@citedi.mx
http://www.citedi.mx

Abstract. Video content is bound to occupy the greater percentage of the bandwidth among its multimedia kin (voice, web, etc.), and also to consume most of the processing resources of the IP-based networks carrying it [1]. Among video streaming applications (compressed video, web seminars, video clips, etc.), videoconference exhibits real time constraints, which sets tight time delivery and packet loss bounds to the network. These constraints are stressed in the mobile environment, where mobile terminal handoff between cells and break-up prone radio links put additional burden to the Quality of Service (QoS) mechanisms of the mobile network. In this work, a computer model of a UMTS release 5 IP core network is developed for the analysis of time delay, packet loss and traffic handling capacity of videoconference sessions. Results obtained indicate that the use of Diffserv for QoS provision can aptly handle a wide spectrum of situations.

Keywords: Streaming video, QoS, UMTS, Diffserv.

1 Introduction

New Third Generation (3G) mobile networks allow up to 2 Mbps for a non-moving nomadic terminal. At these data rates it is outright straightforward to transport multimedia services (video, multimedia messages, etc.). Nevertheless, Quality of Service (QoS) for real time applications imposes stringent bounds of delivery delay on the packet network. Video content is in growing demand in fixed networks, which boast ample to wideband channels and medium to high-speed processors; however, a mobile environment is limited by radio spectrum and mobile terminal processing power. 3G mobile systems (UMTS: Universal Mobile Telecommunications Systems, CDMA2000) show added shortcomings, stemming from the radio-link breakage possibility when a cell-to-cell handover or a steep decrease in signal strength occurs. Under these circumstances, it is of capital interest to characterize the videoconference carrying capacity of 3G systems. The main consideration is that if videoconference Quality of Service requirements are met, those of streaming video in general can also

© S. Torres, I. López, H. Calvo. (Eds.)
Advances in Computer Science and Engineering
Research in Computing Science 27, 2007, pp. 281-289

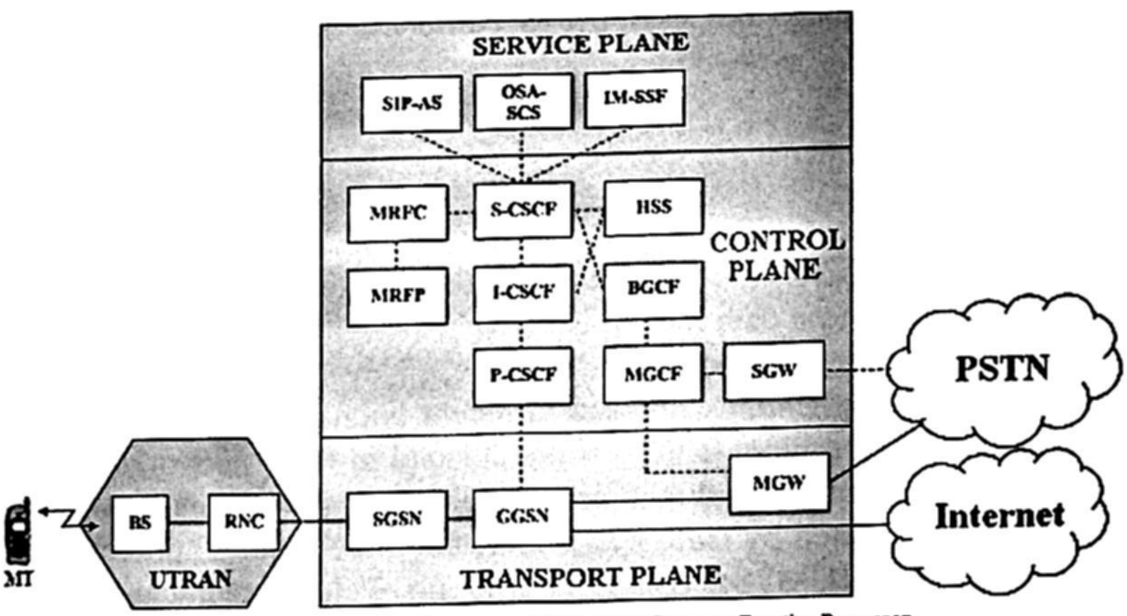
Received 12/02/07
Accepted 08/04/07
Final version 18/04/07

be satisfied. In this context, a UMTS core network model to analyze videoconference stream transport is presented. An H.263 video coder computer model is developed to obtain an accurate behavior of the videoconference data traffic. Time and packet loss constraints are taken care of by a DiffServ discipline in the UMTS IP core network. The effects of resource competition between videoconference and other multimedia streams are also considered.

The organization of the article is as follows. Section 2 explains the main functionalities of the UMTS, especially from the point of view of diverse media transport capacity and QoS provision. In section 3 an UMTS IP Core Network computer simulator is presented, as well as its configuration and operational details. Section 4 deals with the probabilistic representation of the H.263 videoconference traffic behavior. Section 5 puts forth a selected group of simulation outcomes, which help to characterize the limitation and capabilities of the UMTS IP Core Network. Finally, some conclusions are drawn up.

2 3G Mobile System Architecture

Figure 1 illustrates the UMTS release 5 architecture, as defined by the 3GPP (Third Generation Partnership Project). Doted lines represent signaling links, while solid lines indicate bearer channels. The service plain administers and executes the IP based multimedia services. The control plain performs the signaling functions necessary for session management. The transport plain provides the network resources to carry and route the users' data payload.



BGCP: Breakout Gateway Control Function
BS: Base Station
CGSN: Gateway GPRS Support Node
GPRS: General Packet Radio Service
HSS: Home Subscriber Server
IM-SSF: IM - Service Switching Function
I - CSCF: Interrogating - CSCF
MGW: Media Gateway
MGCP: Media Gateway Control Function
MRFC: Media Resource Function Controller
MRFP: Media Resource Function Processor
MT: Mobile Terminal
OSA - SCS: Open Service Access - Service Compability Server
P - CSCF: Proxy - CSCF
RNC: Radio Network Controller
S - CSCF: Serving - CSCF
SGSN: Serving GPRS Support Node
SIP - AS: Session Initition Protocol - Application Server
SGW: Signaling Gateway

Fig. 1. UMTS Release 5.

There are two main functionalities [2]:

(a) UTRAN (UMTS Terrestrial Radio Access Network): Comprised of the base stations (BS or Node B) and the Radio Network Controller (RNC). Handles all the broadcasting functions to connect the Mobile Terminal (MT) and the Base Station, and takes care of mobility management.
(b) IP Core Network (IPCN): Provides and manages the channels to connect the UTRAN to the Internet and PSTN. It also provides end-to-end QoS guarantees.

Within the IPCN, various packet management mechanisms come into play. In accordance with the IP processing of real time applications, the Real Time Protocol (RTP) services videoconference organizing packets in UDP (User Datagram Protocol) [2] datagrams. The GPRS Tunneling Protocol (GTP) [3], [4], [5] communicates the RNC and the GGSN, setting up tunnels to conceal protocol handling of external networks. DiffServ [6], [7], supports QoS requirements by means of assigning priority levels to each data flow entering the edge router, thus specifying which flow will be processed first, second, and so on, at every router in the session path. Eight bit tag identifiers are inserted in every packet to advice the router processor of the packet priority.

3 UMTS IP Core Network Computer Model

Since we are interested on the effects the IPCN is going to exert over the videoconference traffic, it is the bearer conduits we need to focus on, which leads us to the simplified representation of the UMTS system shown in Fig. 2.

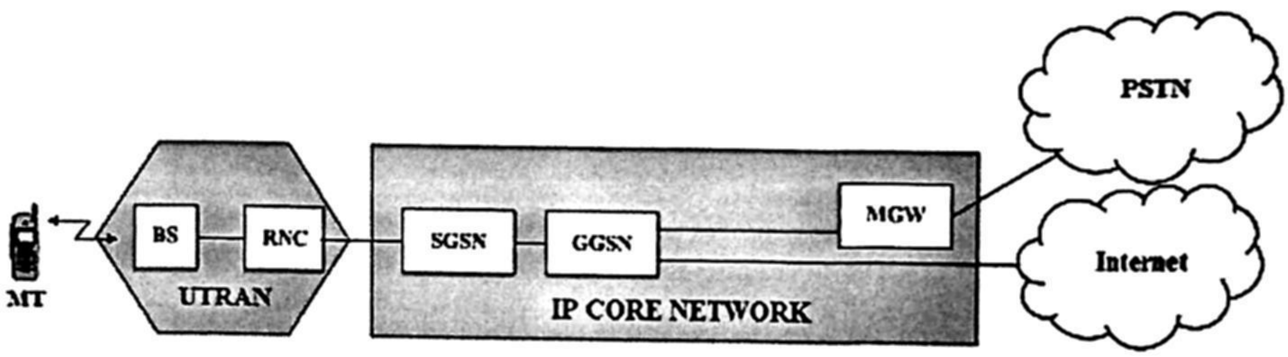


Fig. 2. UMTS simplified core network.

Using this guideline a computer model operating under the Internet Protocol was developed using the OPNET platform [8]. Figure 3 depicts the simulation model wherein each part of the simplified representation of figure 2 can be identified. As an extension to the Internet cloud, a second mobile network is introduced, which contains the complementary videoconference MT. Three additional MTs help shape the multimedia traffic. Two voice sources (MT_Voice) are used; one communicates with the PSTN, and the other with the mobile network B MT. The voice model represents an AMR (Adaptive MultiRate) codec [9] at 12.2 Kbps rate packetized in 20 ms frames (32.5 bytes/frame). Call duration follows an exponential probability density function (t_{av} = 3 min.), with voice activity detection (talk 40%, silence 60 %).

In order to determine the combined effect the network and multimedia traffic will have over the videoconference stream, the MT_Load is configured to yield high volume packetized data streams into the IPCN at various data rates (42 Kbps to 8.4 Mbps) [10]. The MT_Video generates traffic at 66 Kbps average. Its representation and statistical behaviour is developed in the next section.

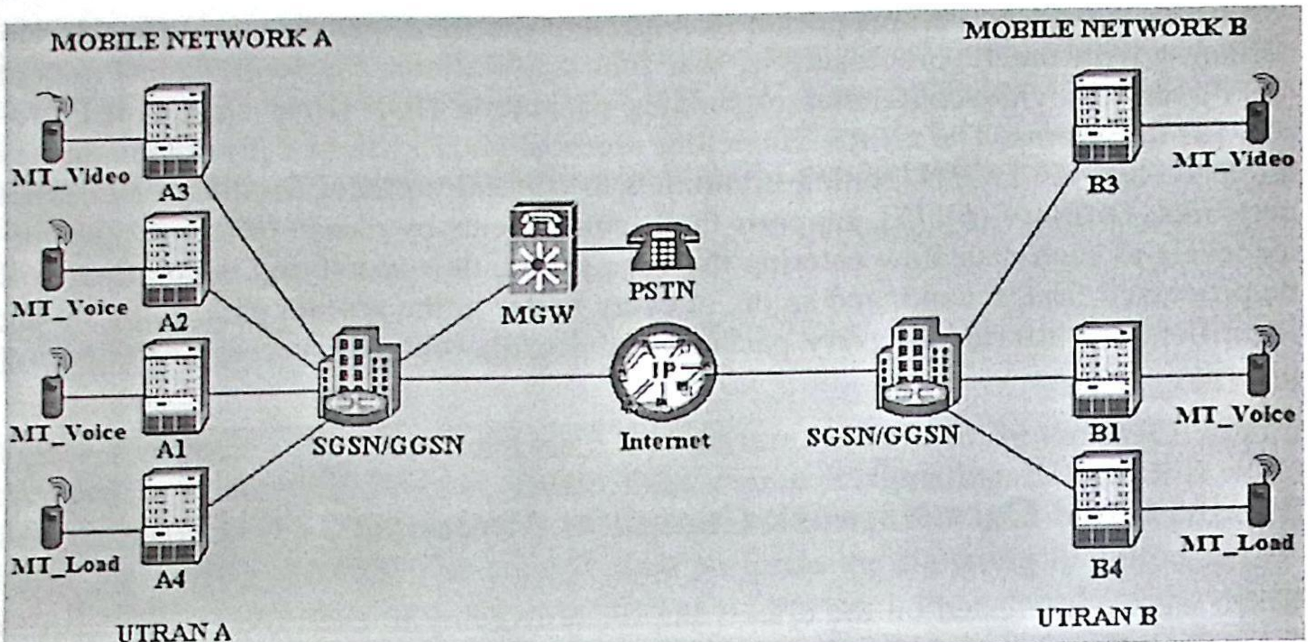


Fig. 3. UMTS-IPCN simulation model.

GTP tunneling is performed in the communications between the RNC and the SGSN/GGSN. The RTP protocol segments the videoconference traffic and introduces sequencing and timing information in each packet [7]. QoS is supplied by the DiffServ protocol, according to the levels assigned in Table 1.

Table 1. DiffServ classes assignment

Traffic Source	DiffServ Class	UMTS Class
UE_Video	AF41 (Gold)	Conversational (Video)
UE_Voice	EF (Premier)	Conversational (Voice)
UE_Load	BE (Best Effort)	Background

4 Videoconference Coder Algorithm and Application

An accurate representation of videoconference data flows was a central requisite for this work, to be able to characterize how the network elements, mechanisms, and other media affect its flow. Several probabilistic behaviour models of coded video data have been developed [11][12][13]. We adopted the Hanzo, et al. [14] model of the H.263 ITU-T (International Telecommunications Union-Telecommunications) [15], [16], recommendation for videoconference. In particular the QCIF (Quarter Common Intermediate Format) format was selected, because its coding range spans from 32 to 384 Kbps, ideal for the bandwidth of 3G radio-channels.

The model is based on a 20 state Markov chain, where each state represents a range of bits/frame of coded video. The probability that k bits/frame will be generated in a specific state is governed by a Poisson probability density function, expressed by

$$\mathrm{P}(k,\lambda_i) = \frac{(\lambda_i T)^k \exp(-\lambda_i T)}{k!} . \quad (1)$$

λ_i is the average number of bits generated in each state, T indicates the time duration of each cycle. Furthermore, λ_i is calculated as follows:

$$\lambda i = R\min + i\frac{R\max - R\min}{N} = R\min + i\Delta R \cdot \quad (2)$$

N is the number of states of the Markov chain; $R_{\max}$ and $R_{\min}$ correspond to the upper and lower limits of the bits/frame respectively. There is also a transition probability from state i to j $\{i = 1, 2\ldots 20\}$. Additional details can be found in [12].

In order to obtain a videoconference stream with an average rate of 66 Kbps the following values were adopted: $T = 10\ \mu s$, $N = 20$ ($0 \le i \le N$), $R_{\max} = 3{,}400$ y $R_{\min} =$ 1,000 bits/frame, $D = 5{,}000$, $O = 3$, $\Delta R = 120$. The algorithm was implemented in Matlab, and run to obtain 3,000 frames, equivalent to 100 secs. of video play. Figure 4 graphics a short window of the results (frames 100 to 200).

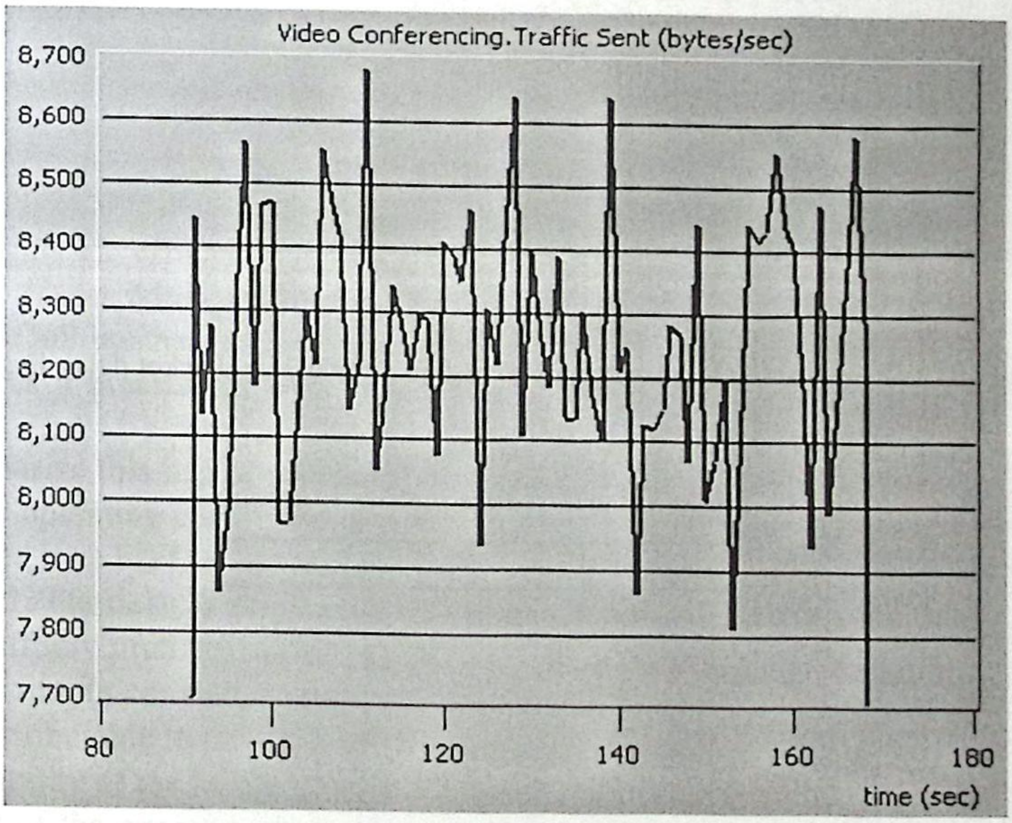


Fig. 4. H.263 coder data rate variation.

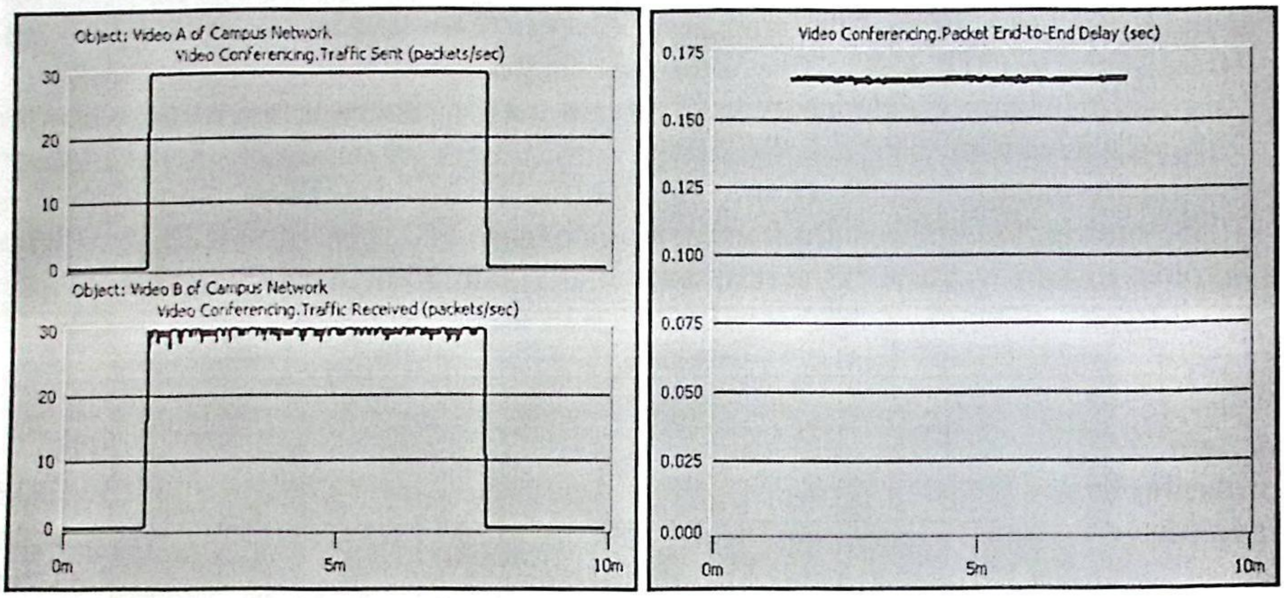
5 Performance Metrics

Over 30 different IPCN operating situations were configured and performed. Table 2 specifies two of them, whose output offers the more relevant information.

Table 2. Core Network settings

	MT_Load data rate	Internet link capacity	SGSN/GGSN processing capability
1	3.5928 Mbps	4.096 Mbps	10.0 Mbps
2	4.4298 Mbps	4.096 Mbps	12.5 Mbps

Figures 5a and 5b illustrate packet loss and delay obtained for operational setting 1, while Table 3 states exact figures for the same setting. Figures 6a, 6b, and Table 4 display corresponding information for setting 2.



a) b)

Fig 5. a) Videoconference stream packet loss in setting 1, b) Packet delay in setting 1

Table 3. Performance metrics in setting 1

Aspect	Value
Session duration	376 s
Packet sent	11277
Packet received	11160
Loss percentage	1%
Maximum delay	165.52 ms
Average delay	164.74 ms

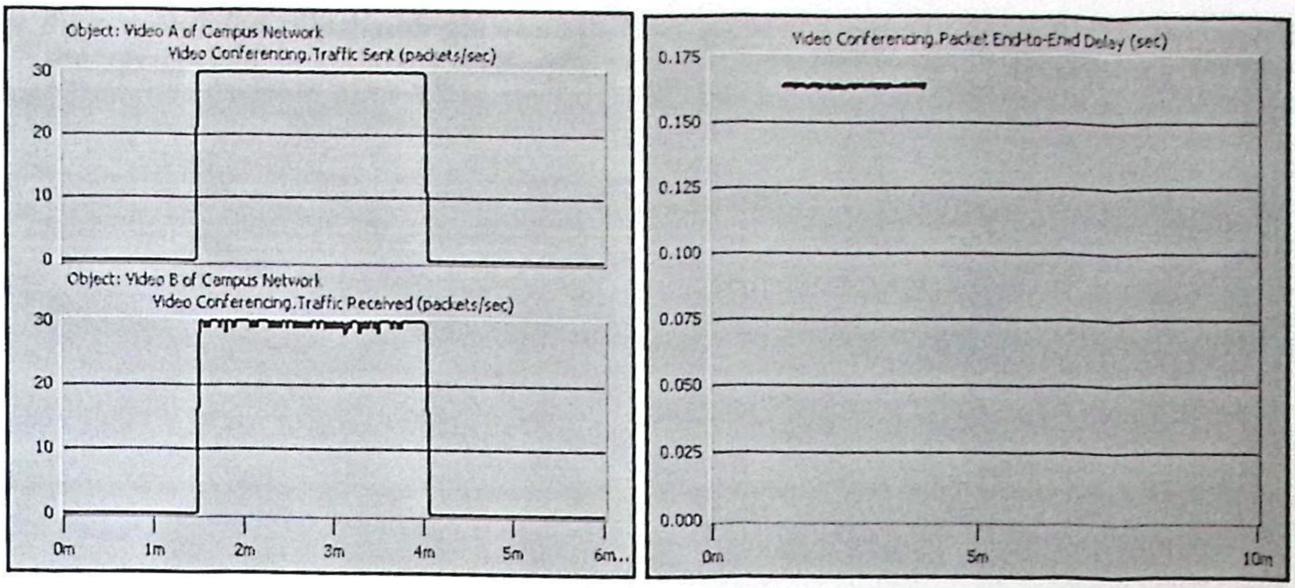


Fig 6. a) Videoconference stream packet loss in setting 2, b) Packet delay in setting 2

Table 4. Performance metrics in setting 2

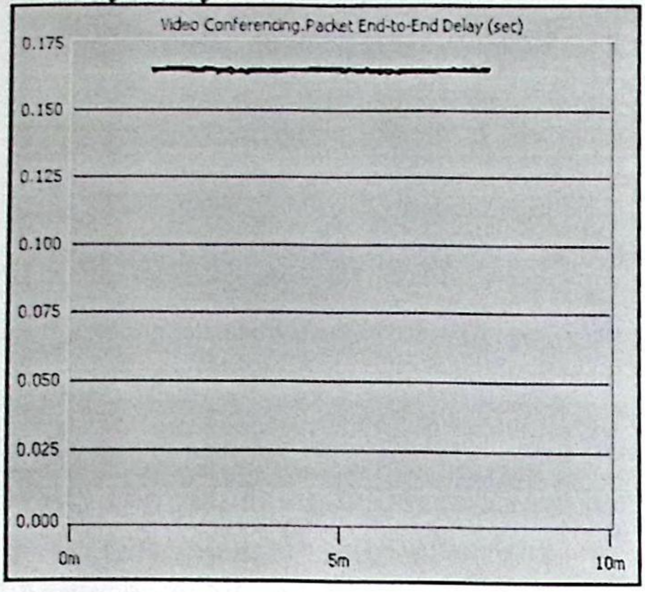
Aspect	Value
Session duration	153 s
Packet sent	4592
Packet received	4545
Loss percentage	1%
Maximum delay	164.53 ms
Average delay	163.80 ms

The outcomes from both settings are very similar in packet loss and time delay, fulfilling the generally accepted constraints: Packet loss ≤ 1%, Delay < 200 msec. This highlights the efficiency of the Diffserv protocol, because in spite that the load in setting 2 causes a congestion condition in the Internet link (4.4298 Mbps > 4.096 Mbps), the QoS mechanism, because of the priority assigned to the videoconference stream, gives it precedence over other traffic, yielding very good performance figures.

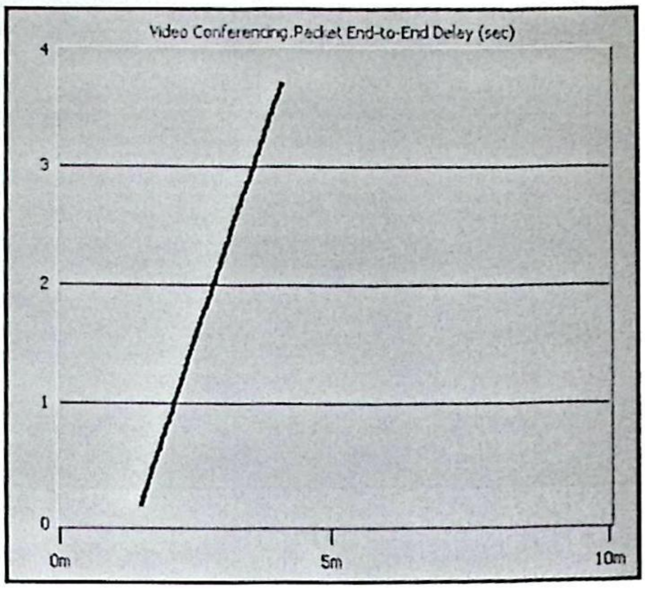
To emphasize this line of reasoning, two graphics are shown below, corresponding to the same operating conditions of Table 2, except for the Diffserv discipline which is substituted by a FIFO (First In First Out), in other words, without QoS support.

In case 1, the delay's range is within boundaries, practically the same obtained with the DiffServ protocol, since the Internet link and SSGN/GGSN router capacities are not overrun. In contrast, a considerable delay in the order of seconds is incurred in setting 2, attributable to the congestion condition mentioned above. The greater processing capability of the SGSN/GGSN router does not help improve the performance.

The rest of the operational configurations tested include a group of more relaxed situations, and other group where the aggregated traffic is roughly double than that of the Internet link or the router capacity, overall showing consistent behaviour with the formerly analyzed.



a)



b)

Fig 7. a) FIFO packet delay in setting 1, b) FIFO packet delay in setting 2

6 Conclusions

Packetized videoconference stands a challenge to any IP network carrying it, because of its low tolerance to delay and packet loss. Thus, it's of capital importance to characterize its behavior, especially in mobile networks.

A computer model of the UMTS Release 5 IP Core Network is developed to obtain key performance parameters of its treatment of coded videoconference, namely, time delay and packet loss. An accurate probabilistic model of the H.263 videoconference stream is also software developed.

The results obtained indicate that the IPCN with Diffserv is capable of handling the videoconference service within operational margins, even in the presence of heavy congestion in the network.

Acknowledgments. This work was financially supported by SIP-IPN grant 2007-0907 and COFAA-IPN Becas por Exclusividad.

References

1. Seeling P., Reisslein M. and Kulapala B., Network Performance Evaluation Using Frame Size and Quality Traces of Single-Layer and Two-Layer Video: A Tutorial, vol 6, IEEE Communications Surveys & Tutorials (2004) 58-78.

2. Camarillo G. and García-Martín M. A., The 3G IP Multimedia Subsystem (IMS): merging the Internet and the cellular worlds, John Wiley & Sons (2004).
3. Wisely D., Eardley P. and Burness L., IP for 3G: Networking Technologies for Mobile Communications, John Wiley & Sons (2002).
4. Chen J. C. and Zhang T., IP-Based Next-Generation Wireless Networks: Systems, Architectures and Protocols, John Wiley & Sons (2004).
5. Kaaranen H., Ahtiainen A., Laitinen L., Naghian S. and Niemi V., UMTS Networks: Architecture, Mobility and Services, John Wiley & Sons (2001).
6. http://www.ietf.org/html.charters/OLD/diffserv-charter.html
7. David Soldani, Man Li and Renaud Cuny, QoS and QoE Management in UMTS Cellular Systems, John Wiley & Sons (2006) 16, 70.
8. http://www.opnet.com
9. 3GPP TS 26.141 v6.0.0 (2004-12), Media formats and codecs. http://www.umtsworld.com/ technology/mediaformats.htm
10. Leal N. J., Modelo de núcleo de red IP de una red 3G para transporte de multimedia en tiempo real, Master's thesis, CITEDI-IPN, México (2006) 66-68.
11. Cosmas J., Petit G., Lehenert R., Blondia C., Kontovassilis K., Casals O., and Theimer T., A Review of voice, data and video traffic models for ATM, European Transactions on Telecommunications, vol. 5, (1994) 139-154.
12. Rose O. and Frater M., A comparison of models for VBR traffic sources in B-ISDN, in Proceedings of the IFIP TC6 Second International Conference on Broadband Communications, (Paris, France), , Chapman and Hall Ltd, London, (1994) 275-287.
13. D Heymann. and Lakshman T., Source models for VBR broadcast-video traffic, IEEE / ACM Transactions on networking, vol. 4 (1996) 40-48.
14. Hanzo L., Cherriman P. J. and Streit J., Wireless Video Communications: Second to Third Generation Systems and Beyond, IEEE Series on Digital & Mobile Communication,. (2001) 205-246.
15. ITU-T, Recommendation H.263: Video Coding for Low Bitrate communication, March 1996.
16. Minoru Etoh, Next Generation Mobile Systems 3G and Beyond, John Wiley & Sons (2005) 247.