This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE GLOBECOM 2005 proceedings.

Analysis of the TCP Round Trip Time over Asymmetric DVB-RCS Systems

Igor Bisio, *Student Member, IEEE*, Mario Marchese, *Senior Member, IEEE*
DIST - Department of Communication, Computer and System Science
University of Genoa, Via Opera Pia 13, 16145, Genoa, Italy
{ igor.bisio, mario.marchese }@unige.it

Abstract—**This paper evaluates the effects of bandwidth asymmetry introduced in a geostationary DVB-RCS satellite architecture on the Round Trip Time of the Transmission Control Protocol (TCP). In particular, it shows the analytical approximation of the round trip time (RTT) of a TCP connection and defines a bandwidth asymmetry index, which is the key point of the work. The final goal is the provision of analytical equations approximating the RTT behavior over asymmetric satellite networks in strict dependence on the asymmetry index. It can help design DVB-RCS system and evaluate their performance. The results are compared with the values obtained through the *ns*-2 simulator.**

Keywords-Satellite Network; Round Trip Time; DVB-RCS; Transmission Control Protocol; Analytical Analysis.

I. INTRODUCTION

Most of the Internet applications currently on the market are build on TCP/IP. On the other hand DVB-RCS (Digital Video Broadcast – Return Channel via Satellite) satellite architectures [1, 2] represent future systems, planned and implemented in Europe, Japan and North America. Matching the applications that use TCP/IP with the need to implement DVB-RCS, it is natural to think of TCP/IP-based applications over satellite DVB-RCS networks. DVB-RCS networks are characterized by bidirectional data transfer between satellite, earth stations and by different channel rates for forward and return channel. Channel capacity asymmetry has a strong impact on the computation of TCP performance models [3].

It is due, in particular, to a rough estimation of the Round Trip Time (RTT). The paper, after describing the characteristic of DVB-RCS architectures relevant for the topic, proposes RTT analytical equations, whose expressions depend on the measure of the asymmetry status summarized by the asymmetry index, also introduced in the paper. The results show that the analytical approximations proposed in closed form may be useful tools for network designing and optimization frameworks.

The paper is structured as follows: section II describes the architecture of a DVB-RCS satellite network. The asymmetry index and the analytical equations of RTT are introduced in Section III. Section IV contains the performance comparison between analytical and simulation measures. Section V lists the conclusions.

II. THE DVB-RCS ARCHITECTURE

A DVB-RCS system allows a two-way data exchange by using satellite terminals. It is composed of: Ground Hub Station, Satellite and Satellite Interactive Terminal (SIT), which is located at the user's site. A user demanding a service (e.g. Internet download or file transfer) makes a login request from a SIT to a Internet Service Provider (ISP) through the satellite channel and the Ground Hub Station connected with the ISP. IP packets flow from the ISP in the opposite direction (Forward Uplink F-U and Forward Downlink F-D). The return links are normally used to carry TCP acknowledgments (Return Uplink R-U and Return Downlink R-D). The forward channel is DVB-S based [1, 2]. From the protocol point of view, the data packets sent by the application layer are transmitted through the TCP/IP suite and encapsulated in a MPE (Multi Protocol Encapsulation) datagram, which adds 16 [bytes]. The last step is the MPEG-2 fragmentation and encapsulation. The MPEG-2 transport packet is composed of a payload of 184 [bytes] and a header of 4 [bytes]. The overall packet size is 188 [bytes]. It is processed and sent to the physical layer. Concerning the return link, two traffic formats may be used: ATM and MPEG-2. The solutions are reported and detailed in [1]. The MPEG-2 format is supposed to be used in this paper. The transmission system (i.e. physical layer) is composed of the following elements: outer coding, inter-leaver, inner coder and QPSK modulator. The outer coder applies a Reed-Solomon code to each MPEG-2 transport packet. The code is fixed and adds a trailer of 16 [bytes]. The bursts are interleaved to reduce the effect of shot errors that may occur over a satellite channel. Finally, before modulation, as reported in [2, 4], a convolutional coder (i.e. the inner coder) adds other redundancy. The system implements convolutional coding with code rates of 7/8, 5/6, 3/4, 2/3, 1/2. Also in the return channel a similar transmission system is provided: Reed-Solomon and convolutional codes may be used. Typical return channel capacities vary from 144 [Kb/s] to 2048 [Kb/s] but this paper considers only small capacities to highlight the asymmetry problems, which may be due not only to the DVB-RCS nature but also to variable code rates. In facts: also with the same values of forward and return bandwidth and buffer size the overall capacity seen by the TCP may be reduced by FEC (Inner Code in particular). It means that also in presence of the same capacity for the forward and return channel a congested bottleneck may be created by both channels in dependence on the status and on the real capacity seen at the transport layer.

III. TCP Round Trip Time Analytical Analysis

A. General Framework

The framework of the analysis is an asymmetric system whose model, reported in Fig. 1 (similarly as in reference [1]), is aimed at emulating the DVB-RCS architecture previously described. A number N of TCP sources is supposed directly conveyed towards the Hub gateway; the return link is used only to carry TCP acknowledgments; an acknowledgment for each transmitted packet is assumed to be sent. The TCP data and acknowledgement packet sizes are constant and equal respectively to l_{data} [bit] and l_{ack} [bit] including TCP and IP headers. The gateways used to access the satellite channel are both characterized by a single buffer with maximum fixed size: $Q_f^{\max}$ data packets in the forward link and $Q_r^{\max}$ acknowledgement packets in the return link. The forward and the return channels are characterized by bandwidth capacities C_f [b/s] and C_r [b/s] and by propagation delays of τ_f and τ_r [s], respectively. The overall bidirectional delay is defined as $\tau = \tau_f + \tau_r$. The channel capacities are supposed directly seen by the TCP layer, thus the bandwidth reduction due to the FEC (Outer and Inner Code) and to the effect of the MPEG-2 encapsulation are included.

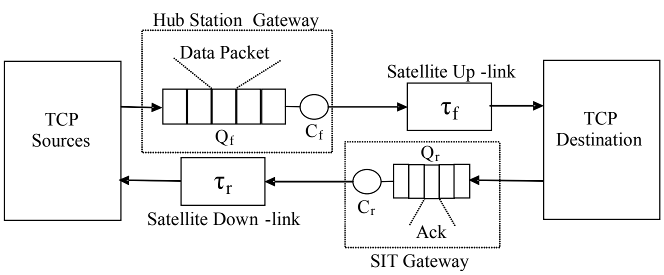

Fig. 1. Asymmetric system simplified model.

B. RTT definitions and asymmetry Index

The TCP Round Trip Time, RTT in the following, of a single packet is defined as the time elapsed from the transmission at the TCP layer of the first bit of a packet to the reception of the last bit of the acknowledgment associated to that packet. RTT values are supposed equal for each TCP source (it implies the fairness condition [5]). The considered parameters are: the propagation delays of the satellite channels (uplink and downlink), the transmission times $1/\mu_f = l_{data}/C_f$ and $1/\mu_r = l_{ack}/C_r$, of data and acknowledgement packets, respectively, and the waiting times in the forward and return buffers (Q_f/μ_f and Q_r/μ_r). Q_f and Q_r are the instantaneous number of packets in the data and acknowledgement queues, respectively, including the packet in service, "seen" by a packet entering the relative queue (the status is checked just before the new packet enters the queue). The round trip time of a data packet entering the queue is shown in (1).

$$RTT = \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{Q_f}{\mu_f} + \frac{Q_r}{\mu_r} \tag{1}$$

The paper is aimed at finding an analytical approximation of the RTT average value, reported in (2).

$$E[RTT] = E\left[\tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{Q_f}{\mu_f} + \frac{Q_r}{\mu_r}\right] \tag{2}$$

Being τ, $\frac{1}{\mu_f}$ and $\frac{1}{\mu_r}$ considered constant over time:

$$E[RTT] = \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{E[Q_f]}{\mu_f} + \frac{E[Q_r]}{\mu_r} \tag{3}$$

A data packet accepted in the system may find ($Q_f^{\max} - 1$) packets in the forward queue and ($Q_r^{max} - 1$) acknowledgements in the return queue in the worst case (i.e., the maximum delay due to the fully occupied buffers). Considering Q_f and Q_r independent, stationary and ergodic stochastic processes, variable in discrete mode from 0 to $Q_f^{\max} - 1$ and from 0 to $Q_r^{max} - 1$, respectively, the RTT mean value may be written as in (4).

$$E[RTT] = \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{1}{\mu_f}\sum_{i=0}^{Q_f^{\max}-1} i \cdot p_{Q_f}^i + \\ + \frac{1}{\mu_r}\sum_{j=0}^{Q_r^{\max}-1} j \cdot p_{Q_r}^j \tag{4}$$

i and j are the number of packets "seen" by an entering packet in the forward and return queue, respectively, while $p_{Q_f}^i$ and $p_{Q_r}^j$ are their related probabilities. The aim is determining (or, for now, approximating) the values of the sums. The first step is the definition of an index to measure the channel asymmetry. Being $\mu_f = C_f/l_{data}$ and $\mu_r = C_r/l_{ack}$ the rates [1/s] of the forward link (packets each second) and of the return link (acknowledgments each second), respectively seen at the TCP layer, the Generic Asymmetry Index α in (5) is defined as the difference between the maximum quantity of data packets that can be stored in the forward channel system (Hub Station Gateway plus satellite Up-link, in Fig. 1) and the maximum quantity of acknowledgement packets that can be stored in the return channel system (SIT Gateway plus Satellite Down-link, in Fig. 1).

$$\alpha = \left(\mu_f \cdot RTT + Q_f^{\max}\right) - \left(\mu_r \cdot RTT + Q_r^{\max}\right) \tag{5}$$

Supposing $Q_f^{\max} = Q_r^{\max}$ (i.e., the maximum number of data and acknowledgment packets that can be memorized in the two respective buffers is the same), the threshold α may be written as:

$$\alpha = \left(\mu_f - \mu_r\right) \cdot RTT = \gamma \cdot RTT \quad \text{where} \quad \gamma \triangleq \left(\mu_f - \mu_r\right) \tag{6}$$

The threshold γ, called Limited Asymmetry Index, "measures" the bandwidth symmetry status. The proposed model of RTT depends on the congested element, which is the forward channel system, if $\gamma \le 0$, and the return channel system, if $\gamma > 0$.

1) *"Return Channel Congestion (RCC)" bandwidth asymmetry* ($\gamma > 0,\ \mu_f > \mu_r$)

This condition happens when the possible congestion element (i.e. the bottleneck) is the return channel. Even if $\mu_f > \mu_r$, the buffer occupancy in the forward buffer is not negligible and both forward and return channel systems contribute to the composition of the round trip time. In practice, both the sums in (4) have to be considered. Actually, at the beginning of each connection, both the forward and the return system contains packets. Even if the return system is the congested element, when an acknowledgment packet is lost, the return channel system is saturated but the data packet transmission is not immediately slowed down by the congestion control algorithm of the TCP protocol. Only when the loss detection occurs (due to duplicated acknowledgments or time out expiration), TCP sources decrease their sending rate. So, when $\gamma > 0$, some packets are simultaneously stored both in the forward and return buffer, contributing to the round trip time, and the equation to be used is reported in (4). The bandwidth asymmetry condition is called *RCC* because the possible bottleneck condition (which depends on the traffic entering the system) may be created only by the return channel.

2) *"Forward Channel Congestion (FCC)" bandwidth asymmetry* ($\gamma \le 0,\ \mu_f \le \mu_r$)

This condition happens when the possible congestion element is the forward channel and, now, the return buffer does not play any role on the round trip delay because it is statistically empty. After a transitory period, in which both the forward and the return system contain packets, when there is loss due to forward buffer overflow, TCP congestion control algorithms slow down the sending rate of the sources while the return system serves all the acknowledgment packets in the queue. So, each new packet entering the system sees the return buffer always empty on statistical basis. In other words: a data packet accesses the forward channel (buffer plus channel) and reaches the destination, its related acknowledgement packet enters the return channel system, finds the buffer empty and is immediately served. The behavior described is mathematically verified when the average time spent in the forward channel system (comprehensive of the propagation delay) is larger than the average time required to empty the return buffer out. From the analytical point of view, it means that:

$$\frac{E\left[Q_f\right]}{\mu_f} + \tau_f + \frac{1}{\mu_f} \ge \frac{E\left[Q_r\right]}{\mu_r} \tag{7}$$

The condition contained in (7), after some algebraic passages, may be expressed as:

$$E\left[Q_r\right] \le \frac{\mu_r\left(E\left[Q_f\right] + \tau_f \mu_r + 1\right)}{\mu_f} \tag{8}$$

Being $\mu_f \le \mu_r$ ($\frac{\mu_f}{\mu_r} \le 1$) and, as a consequence, $E\left[Q_r\right] \le E\left[Q_f\right]$, because the acknowledgement stochastic process entering the return buffer is governed by the packet stochastic process outgoing the forward buffer and the latter is limited by the packet flow entering the forward buffer where packets are possibly lost, it is true that:

$$E\left[Q_r\right] \le E\left[Q_f\right] \le \frac{\mu_r\left(E\left[Q_f\right] + \tau_f \mu_r + 1\right)}{\mu_f} \tag{9}$$

The condition (8) is verified. As said above, it means that, in average, return buffer is empty. The equation to approximate RTT is reported in (10).

$$E\left[RTT\right] = \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{1}{\mu_f}\sum_{i=0}^{Q_f^{max}-1} i \cdot p_{Q_f}^i \tag{10}$$

C. RTT approximations

The computation of $E\left[RTT\right]$, both in (4) and in (10), is a very difficult job because it implies the knowledge of the buffer occupancy probability distributions. This paper proposes three simple assumptions whose effect is checked in the performance evaluation section.

- Maximum Occupancy Assumption (MOA)

Both buffers are supposed completely filled in average:

$$p_{Q_f}^i \cong \begin{cases} 1 & \text{if } i = Q_f^{max} - 1 \\ 0 & \text{otherwise} \end{cases} \quad \text{and} \quad p_{Q_r}^j \cong \begin{cases} 1 & \text{if } i = Q_r^{max} - 1 \\ 0 & \text{otherwise} \end{cases} \tag{11}$$

which implies

$$\sum_{i=0}^{Q_f^{max}-1} i \cdot p_{Q_f}^i \cong Q_f^{max} - 1 \quad \text{and} \quad \sum_{j=0}^{Q_r^{max}-1} j \cdot p_{Q_r}^j \cong Q_r^{max} - 1 \tag{12}$$

hence

$$E\left[RTT\right] = \begin{cases} \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{Q_f^{max}-1}{\mu_f} + \frac{Q_r^{max}-1}{\mu_r}, & \text{if } \gamma > 0 \\ \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{Q_f^{max}-1}{\mu_f}, & \text{if } \gamma \le 0 \end{cases} \tag{13}$$

- Half Occupancy Assumption (HOA)

Both buffers are supposed half filled in average:

$$E\left[RTT\right] = \begin{cases} \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{Q_f^{max}-1}{2\mu_f} + \frac{Q_r^{max}-1}{2\mu_r}, & \text{if } \gamma > 0 \\ \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r} + \frac{Q_f^{max}-1}{2\mu_f}, & \text{if } \gamma \le 0 \end{cases} \tag{14}$$

- No Occupancy Assumption (NOA)

Only propagation and service rates are considered:

$$E\left[RTT\right] = \tau + \frac{1}{\mu_f} + \frac{1}{\mu_r},\ \forall\ \gamma \tag{15}$$

IV. NUMERICAL RESULTS

The RTT analysis proposed is compared with measures obtained through the *ns*-2 simulator. The tests have been carried out by varying the FEC code rate applied as Inner Code at the forward side of the satellite network so varying the capacity really available at the TCP layer and creating different asymmetry situations. The code rate considered are 7/8, 5/6, 3/4, 2/3, 1/2. The Outer Code is always applied. The encapsulation taken as reference, in both the forward channel and the return channel, is the MPEG-2 briefly described in Section II. No channel coding is applied at the return side. The channel capacity at the physical layer is fixed at 500 [Kb/s] then reduced at the TCP layer by FEC code rate (Outer Code and Inner Code) and encapsulation format. For the return link two cases are considered. Case 1: the channel capacity is 20 [Kb/s]; case 2: the channel capacity is 10 [Kb/s]. In both cases the capacities are reduced at the TCP layer by MPEG-2 encapsulation. The application layer implements an FTP server, directly connected to the satellite Hub gateway, which opens 5, 10 and 20 simultaneous active connections during all the simulation time. The data packet length at the data link layer is 1000 [bytes], the ACK size is 40 [bytes] (i.e., the packet lengths before MPEG-2 fragmentation and encapsulation). The maximum buffer lengths are fixed and equal to $Q_f^{\max} = Q_r^{\max} = 11$ [packets] (10 packets in the buffer size plus the packet in service) and $Q_f^{\max} = Q_r^{\max} = 21$ [packets] in two different sets of tests. The propagation delay is 260 [ms] for both forward and return channel coherently with geostationary satellite environments. TCP settings are fixed and the Reno version is taken as reference. Figures from 2 to 7 contain the mean buffer occupancy in cases 1 and 2. For each code rate of the case 1 the Limited Asymmetry Index is negative: the return buffer condition expressed in (7) is easily verified and only the forward buffer is traversed by packets, as clear in the figures. The mean buffers occupancy is about the half of the overall buffer size when the number of TCP sources is 5, thus the HOA approximation is reliable in this case. When the number of TCP active sources grows, the mean buffer occupancy increases and the suitable approximation is MOA. NOA approximation is less applicable in all cases. In case 2 the Limited Asymmetry Index is positive when the code rate is larger than ½: both the forward and return buffers contribute to compose RTT, as clear in Figures from 2 to 7. When the code rate is ½ and the Limited Asymmetry Index is negative, the occupancy of the return buffer component is again negligible. The average forward buffer occupancy is about half of the overall buffer size when the number of TCP sources is 5 but tend to a larger value close the maximum size when the number of sources increases. In this case both the HOA and MOA approximations are satisfactory. NOA, also in this case, is very rough. The RTT mean value behavior is shown from Fig. 8 to Fig. 11. The simulation results (the mean value of the RTT averaged over the number of TCP sources), indicated as SIM in the figures are compared with the approximations proposed in the paper. HOA and MOA have the same slope of the simulation results, and they are close to the measured values in both case 1 and 2. MOA is always conservative while HOA is fully reliable only for case 1 and buffer set to 10.

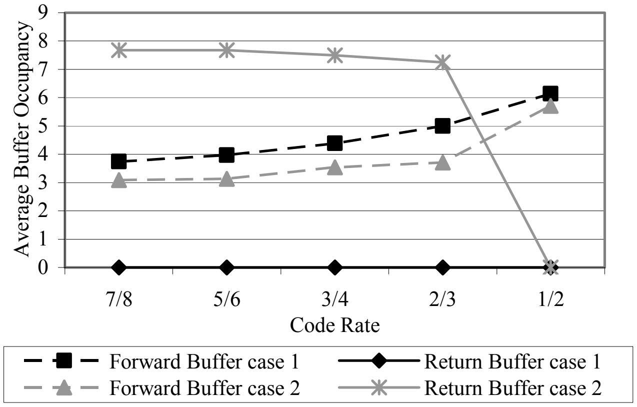

Fig. 2. Average buffer occupancy (5 TCP sources, buffer size 10).

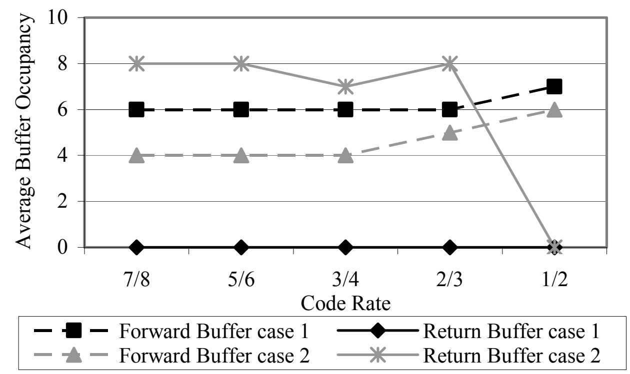

Fig. 3. Average buffer occupancy (10 TCP sources, buffer size 10).

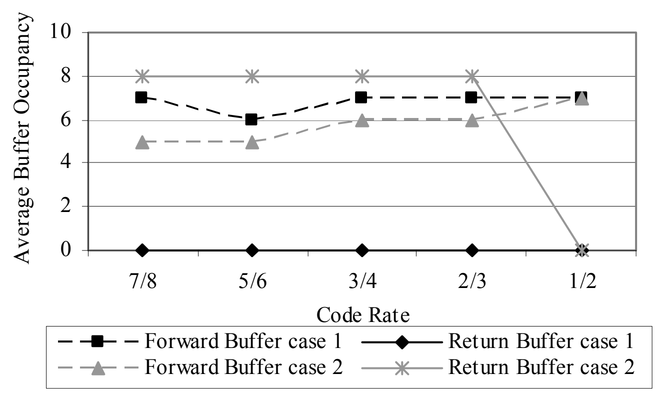

Fig. 4. Average buffer occupancy (20 TCP sources, buffer size 10).

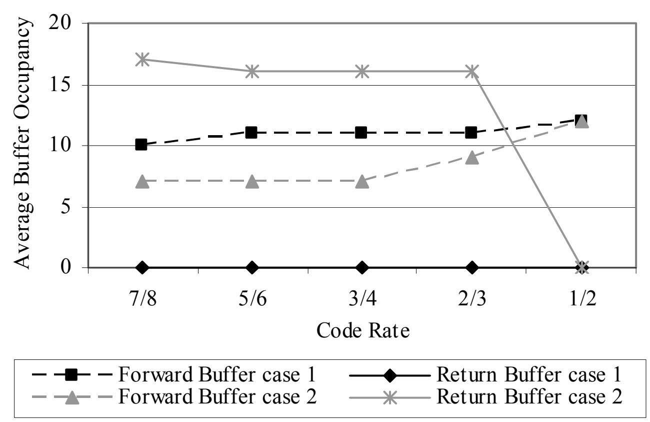

Fig. 5. Average buffer occupancy (5 TCP sources, buffer size 20).

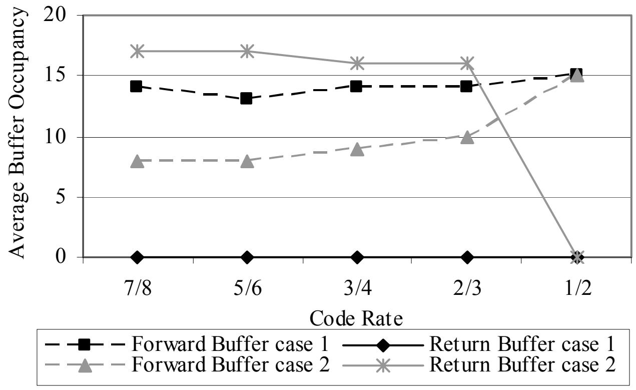

Fig. 6. Average buffer occupancy (10 TCP sources, buffer size 20).

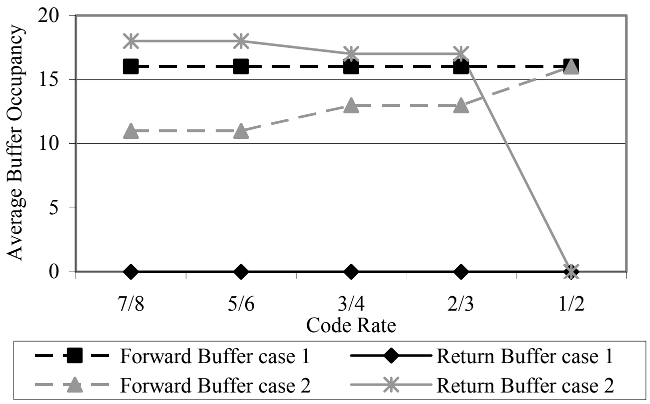

Fig. 7. Average buffer occupancy (20 TCP sources, buffer size 20).

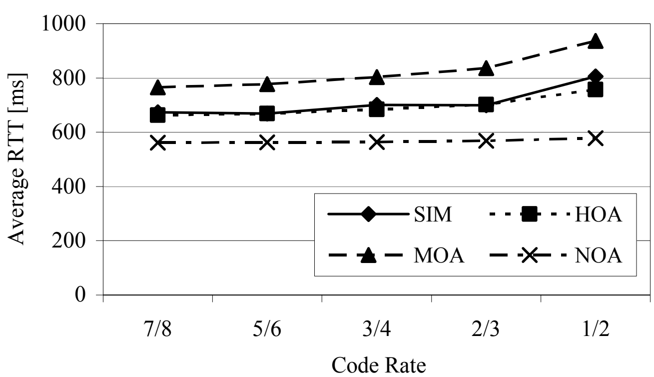

Fig. 8. Simulation and Analysis of the Round Trip Time (Case 1, buffer size 10).

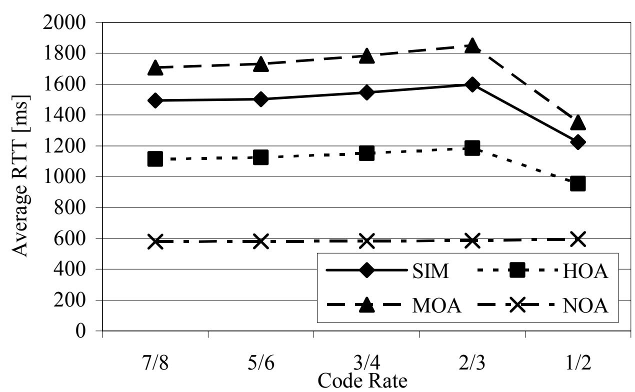

Fig. 9. Simulation and Analysis of the Round Trip Time (Case 1, buffer size 20).

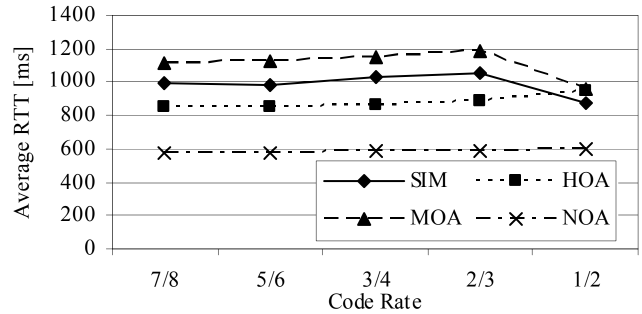

Fig. 10. Simulation and Analysis of the Round Trip Time (Case 2, buffer size 10).

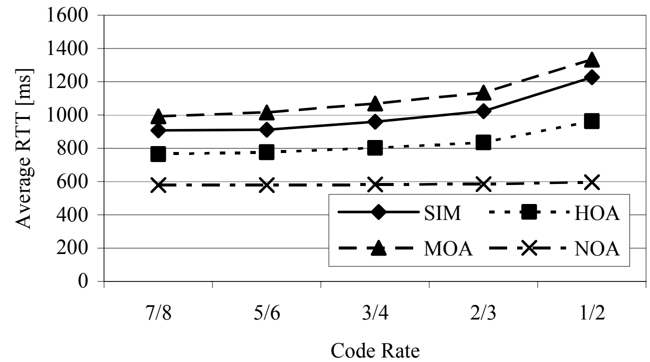

Fig. 11. Simulation and Analysis of the Round Trip Time (Case 2, buffer size 20).

V. CONCLUSIONS AND FUTURE WORKS

Analytical approximations of the mean value of the RTT, as a function of the available resources (i.e. bandwidth and buffer) has been introduced for a DVB-RCS asymmetric satellite system. A simple analytical measure of the channel asymmetry (Limited Asymmetry Index), has been defined and used as key point of the work. The analytical results are compared with measures obtained through the *ns*-2 simulator and show a good degree of accuracy. The accurate analysis of the RTT by using precise expressions of its components instead of approximations also considering alternative transport layers suited for satellite environments will be the future direction of the study.

REFERENCES

[1] Digital Video Broadcasting (DVB), "Interaction Channel for Satellite Distribution Systems," *ETSI Standard EN 301 790*, June 2003.

[2] Digital Video Broadcasting (DVB). "Framing Structure, Channel Coding and Modulation for 11/12 GHz Satellite Services," *ETSI Standard EN 300 421*. November 1997.

[3] T. V. Lakshman, U. Madhow, B. Suter, "TCP/IP Performance with Random Loss and Bidirectional Congestion," *IEEE/ACM Transactions on Networking*, vol. 8, no. 5, October 2000, pp. 541-555.

[4] D. Cammia, L. Favalli, "Turbo-Codes and Layered Video for Adaptive QoS and Improved Service in the DVB-S System," *IEEE Global Telecommunications Conference*, November 2001, San Antonio, Texas, USA, pp. 2050-2054.

[5] D. Chiu and R. Jain, "Analysis of the Increase/Decrease Algorithms for Congestion Avoidance in Computer Networks," *J. of Computer Networks and ISDN*, vol. 17, no. 1, June 1989, pp. 1-14.