International Journal of Scientific & Engineering Research, Volume 6, Issue 3, March-2015 427
ISSN 2229-5518

Evaluation of end to end delay and packet loss probability in congestion control system

OJEDAYO, Benson, OYINLOYE, Elohor

1. ABSTRACT

In recent years there has been an extensive growth in the number of Internet users, hosts and applications. The success in coping with the fast growth of the Internet rests on the IP (Internet Protocol) architecture's robustness, flexibility, and ability to scale the underlying end-to-end paradigm of the IP architecture.

This architectural principle is embodied in the main transport protocol of the Internet, TCP (Transport Control Protocol) . This has proven to be crucial to the success of the Internet.

This rapid growth of the Internet and the proliferation of its new applications pose a serious challenge in network performance management and monitoring. The sheer volume of network traffic imposes a burden on network administrators, and demands a visual interface for easy grasp of the current status of the network. The ever-expanding topology renders debugging a very complex task. In this research, the issues concerning measurement and analysis of end-to-end performance, more specifically, delay and loss, using active measurement are addressed.

Keyword: Congestion control, end-to-end delay, open loop, closed loop, Round Robin Technique, Network Packet

1.0 INTRODUCTION

Congestion control is the state of network overload. In networking, it is a situation in which a switch or router has so many packets queued for transmission so that it runs out of buffer space and must start dropping packets in the case more arrive. Congestion control systems could either be open loop congestion control or closed loop congestion control systems. One of the traffic control mechanisms is buffering for the sharing of access to bandwidth capacity. Buffering is used to share out the available rate in the sessions and where the aggregate arrival rate can exceed the service rate for significant period of time [1]

2.0 SYSTEM ANALYSIS AND DESIGN

End-to-end delay traces are frequently used in analysing network performance. The accuracy of such measurements depends on the time involved and how the measurement is synchronized.

To obtain an accurate measurement of one-way delay, errors and uncertainties related to time must be accounted for. When one of the clocks involved in the measurement resets its time, the measured delay using the timestamps from two clocks may be affected, depending on the comparative magnitudes of delay and the time adjustment.

- OJEDAYO, Benson is a lecturer with the Computer Engineering Department, Ekiti State University, Ado–Ekiti, Nigeria. bentheta@yahoo.com
- OYINLOYE, Elohor is a lecturer with the Computer Science Department, Ekiti State University, Ado–Ekiti, Nigeria.

The end-to-end delay consists of transmission and propagation delays plus variable queuing delay. When all of the packets go over the same route to the receiver, they incur the same propagation delay, regardless of size; the transmission delay is also the same. Even if the packets go over the same route, and are of the same size, the packets experience different

address problems in delay measurements due to clock adjustments and rate mismatches [2] It uses forward and reverse path measurements of delay between a pair of hosts to deal with clock synchronization problems, such as relative offset and skew. Many applications, however, see only one-way delay (e.g Internet telephony, video-on-demand applications, RealPlayer, web TV), and still have to deal with the clock synchronization problems in packet delay. One-way measurements alone are not enough to infer the clock offset, and we cannot distinguish the clock offset from the fixed portion of end-to-end delay. Seconds is due to the time difference between clocks and the fixed transmission and propagation delay, without the availability of more information. Due to this lack of information in one-way delay measurements, the focus is on the variable portion of one-way delay measurements.

The variable queuing delay serves a very important role in network and application design. Continuous-media applications such as audio and video need to absorb the delay jitter perceived at the receiver for smooth playout of the original stream [3]. Determining the correct amount of buffering, and reconstructing the original timing is crucial to the performance of continuous-media applications [4]. The variable queuing delay is also useful in monitoring network performance at the edges of the network; the transmission and propagation delays are fixed per route, and do not convey any information about the dynamic changes inside the network when packets follow a fixed route.

This research emulates the environment of open and closed loop systems and the end-to-end delay in the arrival of packets sent.

This proposed system has 4 basic benefits, viz

- Easy method of determining end-to-end delay
- Enhances information on how packets are lost
- Formulate a better system that simulate open and closed loop controls
- Time measurement of open and closed loop system

2.1 DESCRIPTION OF PACKETS IN A NETWORK

The figure 3.1 depicts the state transition diagram of a mobile node trying to transmit packets to another mobile node in a mobile Ad hoc Network at the Medium Access Control (MAC) layer. This delay is the difference in time between the moment a packet reaches the head of the queue to the time the sender knows the packet is successfully received through the reception of an ACK. This expression of MAC delay gives average service time of a packet in a node. It consists of three parts:

- Time to transmit packet successfully once
- Total time a node spends in backoff
- Total transmission time used for retransmission of the packet

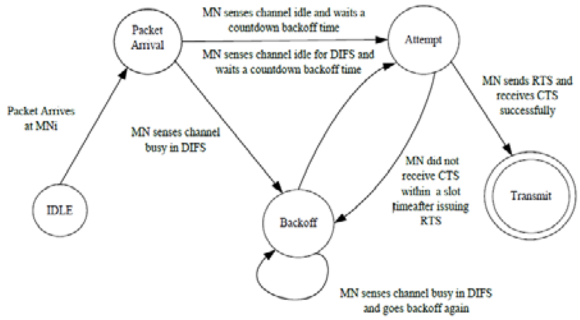

Fig.: 3.1 Packets description

2.2 THE ANALYSIS OF END-TO-END DELAY

End-to-end delay is the delay encountered by a packet which can be measured from the time the packet is generated to the time the source node receives acknowledgement that it was successfully delivered to its destination. Delaying of packets consists of the queuing delays at the intermediate node and MAC delay observed at the source and intermediate nodes.

3.0 SYSTEM DESIGN

There are two phases in the design of the system: the open loop phase and the closed loop phase.

3.1 THE OPEN LOOP SYSTEM PHASE

The open loop systems as shown in fig 3.2 are a non-feedback system in which the control input to the system is determined using only the current state of the system. Once the system is up and running, mid-course corrections are not made.

Processing time for open loop system is given as

$$P_T = {}^{L}/_{T} \qquad (1)$$

Packet loss probability for the system is calculated thus

$P_A = {}^{D}/_{L}$ (2)

Congestion control for open loop system is given as

$C_C = {}^{(P_T)}/_{2} \times P_A$ (3)

Delay time for open loop system is

$\sum(P_T) - \sum(n)$ (4)

Source data transmission is gotten as

$S_D = D_{PT}$ (5)

Where

P_A is the Packet loss probability of the system

D is the delay time

L is the length of packet contents

T is the transmission time

P_T is the processing time

C_C is the congestion control of open loop System

n is normal transmission time

S_D is source data transmission

D_{PT} is the destination processing time

3.2 OPEN LOOP SYSTEM ARCHITECTURE

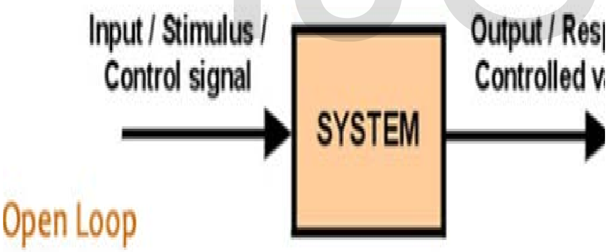

Fig.: 3.2 Open loop system

3.3 THE CLOSED LOOP SYSTEM PHASE

Closed loop congestion control as shown in fig 3.3 is a control mechanism that tries to alleviate congestion after it happens. Several mechanisms have been used by different protocols such as: backpressure, choke packet, implicit signalling, explicit backward signalling, and forward signalling. Closed loop shows a closed-loop action and can counteract against disturbances. Processing time for open loop system is given as

$P_T = {}^{L}/_{T}$ (6)

Packet loss probability for the system is calculated thus

$P_A = {}^{D}/_{L}$ (7)

Congestion rate for closed loop system is

$C_R = P_T - T_T$ (8)

The delay time is

$D = \sum (P_T)^2 - \sum(n)$ (9)

Where

P_T is processing time

L is the length of packet sent

T is transmission frequency

P_A is the packet loss probability

D is the delay time of the system

C_R is the congestion rate of the system

T_T is transmission time of the packet

n this is the normal transmission time

3.4 CLOSED LOOP SYSTEM ARCHITECTURE

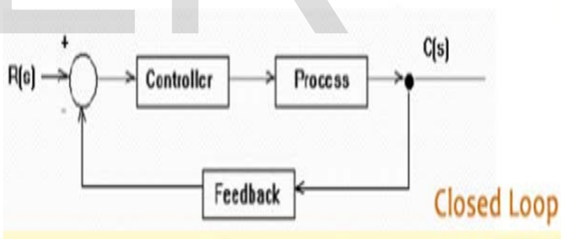

Fig 3.3 closed loop system

One way in which we can accurately control the process is by monitoring its output and "feeding" some of it back to compare the actual output with the desired output so as to reduce the error and if disturbed, bring the output of the system back to the original or desired response. The measure of the output is called the "feedback signal" and the type of control system which uses feedback signals to control itself is called a Close-loop System.

3.5 SIMULATION OF THE SYSTEM

In communication and computer network research, network simulation is a technique where a program models the behaviour of a network either by calculating the interaction between the different network entities (hosts/packets, etc.) using mathematical formulas, or actually capturing and playing back observations from a production network. The behaviour of the network and various applications and services it supports can then be observed in a test lab; various attributes of the environment can also be modified in a controlled manner to assess how the network would behave under different conditions.

The system shows simulation for an open and closed loop system control. The scheduling method used for both simulations is the Round Robin (RR) technique. For the open loop, the packets to be transferred were randomly generated according to the specification given. The time limit for the transfer of each packet was also specified. At the start of transmission, the sender sends a packet to the receiver for processing and display. The time of processing depends on the length of the packet being sent. If the processing time is less than the transmission time, the receiver displays the packet received and indicates its readiness to receive another packet for the same process. On the other hand, if the processing time exceeds the transmission time, a delay occurs because the sender has to wait for the receiver to be free before any new packet is transferred. At the end of that transfer, the delay time for the period is calculated by subtracting the transmission time from the processing time. This prevents congestion. The closed loop on the other hand does not prevent congestion. If the processing time of exceeds the normal transmission time, the transmission time of the sender is increased by the amount of time the receiver needs to process the last packet. Round robin technique is a load balancing technique in which balance power is placed in the DNS server instead of a strictly dedicated machine as other load techniques do. Having a document disseminated from one person to another in a group with persons adding comments can also be used to mean the same thing.

4.0 SYSTEM DEVELOPMENT IMPLEMENTATION

Systems development is the process of creating and maintaining information systems.

The system is developed using scheduling method for both simulations called the Round Robin (RR) technique. The system is implemented using visual basic.NET (VB.NET) that runs on net framework 3.5.

4.1 OPERATION OF THE SYSTEM

The operation of the system is sub-divided into two:

- **INITIAL REQUIREMENTS**

The initial requirements are

(i) Send packet on a network and study the delivery detail.

(ii) Evaluate the delay of the sent packets on the two systems.

- **FUNCTIONAL REQUIREMENTS**

Using the initial requirements as a starting point, there follows a more complete formal set of functional requirements for the system.

(i) Must Extract Detail Of Open And Closed Loop Packets From The Output

The system must be able to indicate if the system is an open or closed loop.

(ii) Must Be Able To Generate Packets

The system must be able to generate a specified number of packets

(iii) Must Be Able to Optionally Control Congestion

In addition to the previous requirement it must be possible to further control the data generated. This should include options to control number of Packets and Delay in response.

(iv) Must Display and Update Loss Probability

The system must be able to dynamically update the loss probability and the delay time as monitoring output is considered

(v) Interface must respect the native appearance of the operating system

The system must render an interface which conforms to the native appearance and layout of the operating system under which it is running.

4.2 PROGRAM IMPLEMENTATION PROCEDURE

4.2.1 OPEN LOOP SYSTEM ENVIRONMENT

If the user chooses the open loop system, the figure 4.1 will be displayed.

The number of packet option gives the user opportunity to choose the number of packet that is to be transmitted (it's normally 5 packet by default), after that, the user can click on load packets to load the packet since we are working on simulation environment the system will automatically generate five data to be transmitted (using round robin techniques) at the sender's end.

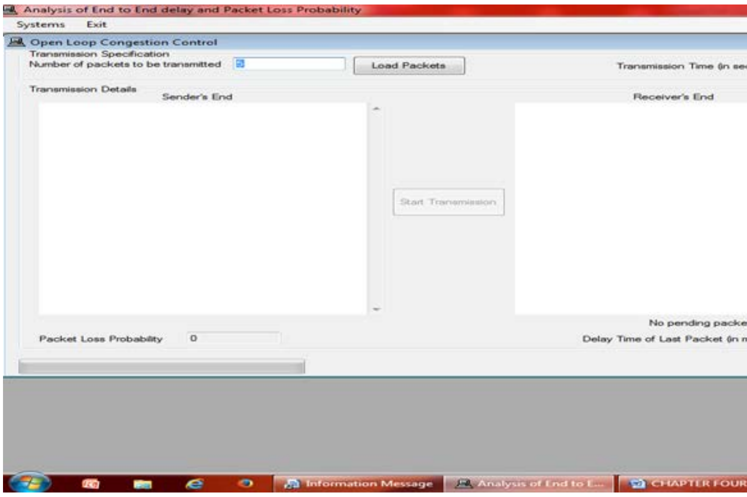

Fig.4.1: open loop system simulation environment

4.2.2 DATA TRANSMISSION PAGE

After the packet to be transmitted is ready, as shown in fig 4.2, the user can click on start transmission bottom to begin the transmission proper. The packet transmitted will be received at the receiver's end and in the process of transmitting it; the system will automatically calculate the packet loss probability and delay time in milliseconds for each packet sent.

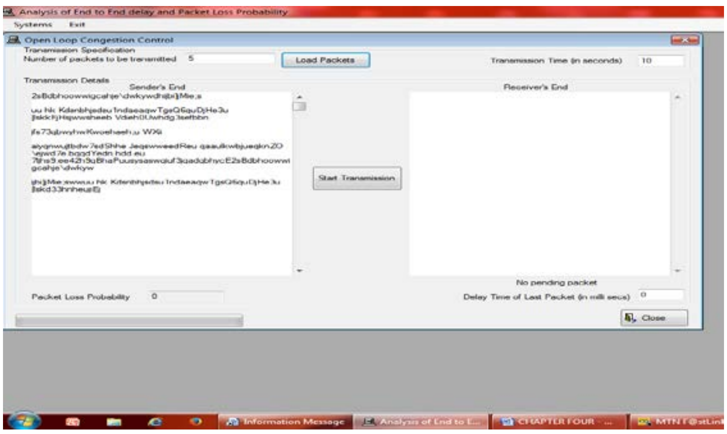

Fig 4.2: showing result of data to be transmitted at the sender's end

4.2.3 PACKET LOSS PROBABILITY AND DELAY TIME PAGE

After the entire packet has finish transmitting a box will be popped up showing that the packet has finished transmitting as shown in fig 4.3.

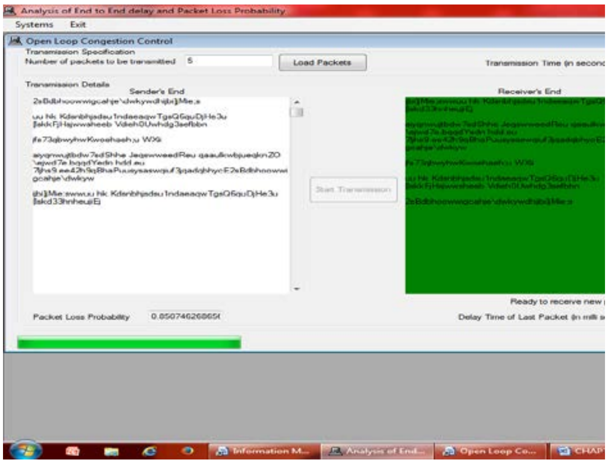

Fig. 4.3: showing the result of packet loss probability and delay time

4.3 CLOSED LOOP SYSTEM SIMULATION ENVIRONMENT

Hence if the user chooses the closed loop system the following will show

Number of packet option give the user opportunity to choose the number of packet that is to be transmitted (it 5 packet by default), after that, the user can click on load packets to load the packet since we are working on simulation environment the system will automatically generate five data to be transmitted (using round robin techniques) at the sender's end.

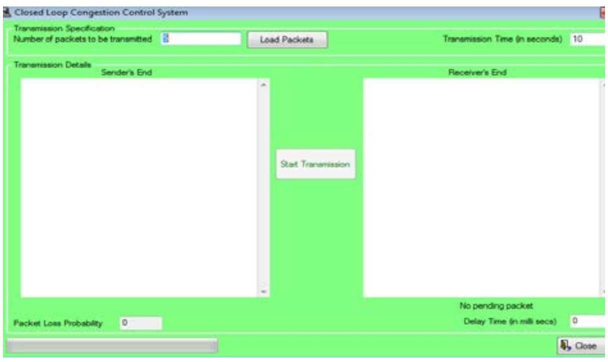

Fig 4.4: closed loop system simulation environment

4.3.1 DATA TRANSMISSION PAGE

After the packet to be transmitted is ready, the user can click on start transmission bottom to

Number of packets to be transmitted	Transmission Time (in seconds)	Delay Time (in milliseconds)	Probability Loss
5	10	22	0.688
10	10	37	0.787
15	10	92	0.902
20	10	141	0.934

begin the transmission proper. The packet transmitted will be received at the receiver's end and in the process of transmitting it; the system will automatically calculate the packet loss probability and delay time in milliseconds for each packet sent

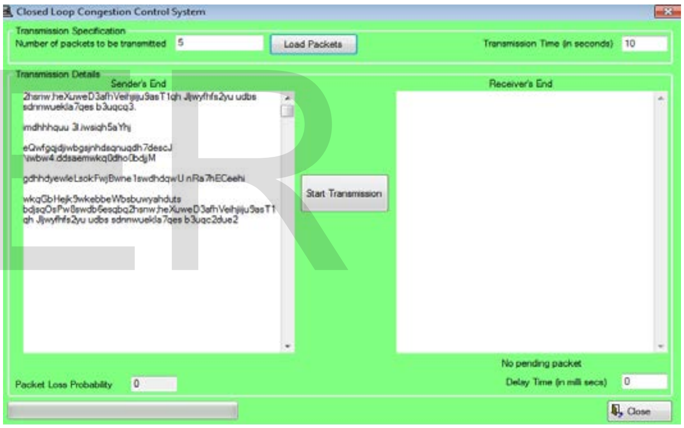

Fig 4.5: showing result of data to be transmitted at the sender's end

4.3.2 PACKET LOSS PROBABILITY AND DELAY TIME IN CLOSE LOOP

After the entire packet has finish transmitting a box will be popped up showing that the packet has finished transmitting

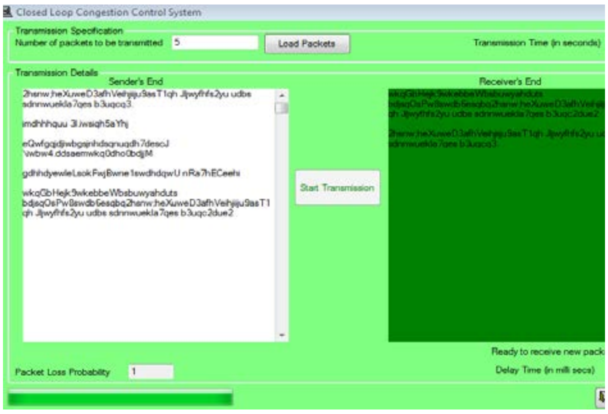

Fig. 4.6: showing the result of packet loss probability and delay time

TABLE 4.1 SHOWING RESULT OF OPEN LOOP SYSTEM

TABLE 4.2 SHOWING RESULT OF CLOSED LOOP SYSTEM

4.4 EXPERIMENTAL ANALYSIS

The open and closed loop system developed is designed to analyse the end-to-end delay and packet loss probability. The testing was performed using four different transmitted packets with a constant time (10 seconds) on the two systems (open and closed loop control systems). It can be seen from table 4.1 and Table 4.2 that the larger the number of packet the more the delay time in both system. The probability loss of closed loop system is higher than open loop system.

5.1 SUMMARY/CONCLUSION

The research work shows the analysis of end-to-end delay and packet loss probability in congestion control system. The model was developed using development tools that makes it easy to analyse packet and obtain the delay time. This application promotes the use of simulation which helps in the conservation of time when applied to real system. The research has resulted in a functional application which achieves the goals set and attain a high level of adherence with the requirements specified.

The research work which is on simulation of open and close loop system, it was able reveal that there is bound to be delay in the delivery of a sent packet and it is dependent on the number of packet sent and the time involved. This study explains well that the introduction of the simulated network in the prediction of the real system gives a clear picture of how packets are sent and received in the two systems and the observation is that open loop system is faster than closed loop system because it prevents congestion from occurring meanwhile the packet loss probability of open loop system is lower than closed loop system.

Number of packets to be transmitted	Transmission Time (in seconds)	Delay Time (in milliseconds)	Probability Loss
5	10	38	1
10	10	82	1
15	10	133	1
20	10	186	1

REFRENCES

[1] Roberts, J.W., (1996) "QoS pricing and admission control in the internet". Retrived from http://www.rennes.enst-bretagne.fr/RHDM99/QoSInternet on 1996.

[2] Paxson, Vern. (1997), "Measurements and Analysis of End-to-End Internet Dynamics". University of California, Berkeley.

[3] Ramjee, R., Kurose, J., Towsley, D., Schulzrinne, H. (1994), "Adaptive playout mechanisms for packetized audio applications in wide-area networks". In Proceeding of INFOCOM '94.

[4] Moon, S.B., Kurose, J., and Towsley, D. (1998), "Packet audio playout delay adjustment: Performance bounds and algorithms. ACM/Springer Multimedia Systems 5 January, 1998.

[5] Paxson, V. (1998), "On calibrating measurements of packet transit times: In Proceedings of SIGMETRICS '98 (Madison, Wisconsin, June 1998).

[6] Bitorika, A (2004), "A comparative study of active queue management schemes", Proc. of IEEE ICC Congestion Control under Dynamic Weather Condition, Ireland.

[7] Brandauer,C (2001), "Comparison of tail drop and active queue management performance for bulk- data and web-like Internet, Proc. of IEEE ISCC 2001, pp.122-129

[8] S. Tanenbaum, (2002), "Computer Networks", 4th Edition, Prentice Hall Ptr.

[9] Bernet, Y., (2000) "The complementary roles of RSV and differentiated services in the full-service QoS network", IEEE Communications Magazine, February, 2000.

[10] F. Wu-chang, (2002) "The blue active queue management algorithms", IEEE/ACM Transactions on Networking, vol.10, pp.513-528.

[11] G. Thiruchelvi and J. Raja, (2008) "A survey on active queue management mechanisms", IJCSNS International Journal of Computer Science and Network Security, vol.8, 2008.

[12] Haverkort, B.R., (1998) "Performance of Computer communications systems", Wiley 1998.

[13] Holness, F. (2000) "Congestion control mechanisms within MPLS", PhD, Thesis University of London, 2000.

[14] Hussmann, H., Mamais, G., Prehofer, C., Salsano, S., Eichler, G. (2000) "Implementing Integrated and Differentiated Services for the Internet with ATM networks" a practical approach, IEEE Communications Magazine January.

[15] J. Aweya (2011), "A control theoretic approach to active queue management", Computer Networks, vol.36, pp.203-235.

[16] Jon, Postel. (1981), "Transmission control protocol specification". RFC 793, Internet.

[17] Jung, J. (1996), "Quality of service in telecommunications", Part 1. IEEE Communications Magazine, August, 1996.

[18] Kerman, P. (1977), "Switching and flow control techniques in computer communication networks", PhD dissertations, computer science department UCLA.

[19] Knightly, E.W., Schroff, N.B (1999), "Admission control for Statistical QoS: Theory and Practice", IEEE Network journal, March, 1999.

[20] Kurose, J. (1993), "On per session delay distributions and the call admission problem for realtime applications, conference proceedings on communications, architectures, protocols and applications". Sept 13-17, 1993, San Francisco, Calfornia, USA.

[21] Kurose, J.K., Mouftah, H.T. (1998), "Computer-aided modelling, analysis and design of communication

networks", IEEE J. Sel. Areas Communication, SAC-6, (1), pp.130-145.

[22] Liebeherr, J. (2000) "Buffer Management and Scheduling for Enhanced Differentiated Services", Technical Report, University of Virginia.

[23] Liebeherr, J., Patek, S., Yilmaz, E. (2000) "Tradeoffs in Designing Networks with End-to-End Statistical QoS Guarantees", IEEE/IFIP International Workshop on QoS' 2000, June 5-7, 2000.

[24] Management", AT&T Center for Internet Research, 2001.

[25] Mills, D. (1992), "Network time protocol (version 3): specification, implementation and Tech. rep., Network Information Center, SRI International, Menlo Park, CA, March 1992.

[26] Nicol, D.M. (1998), "Special Issue on the Telecommunications Description Language", ACM Sigmetrics Performance Evaluation, Vol. 25, no 4, Mar. 1998.

[27] Onvural, R.O. (1994), "Asynchronous Transfer Mode Networks", Performance Issues, Artech House,INC.

[28] Pitts, J.M. (1993), "Cell-rate simulation modelling of Asynchronous Transfer Mode telecommunication networks" Ph.D Thesis University of London.

[29] Pitts, J.M. "Schormans, J.A. (2000), "Introduction to IP and ATM design and performance", 2nd edition, Wiley.

[30] Pitts, J.M., Schormans, J.A. (1996) "Introduction to ATM design and performance", Wiley.

[31] Pitts, J.M., Schormans, J.A., Pearmain, A.J. (2001) "Tight bounds on the tail of the packet waiting time distribution in buffered networks". International Journal on Communications, Vol 14, Issue Published by John Wiley and Sons.

[32] Price, W.L. (1977) "Data network simulation experiments" at National physical lab 1968-1976 computer networks, vol. 1.

[33] R. Stanojevic et al (2006), "Adaptive tuning of drop-tail buffers for reducing queuing delays", IEEE, Com-Communications Letters, vol.10, pp.570-572.

[34] Rakocevic, V. (2002) "Dynamic Bandwidth Allocation in Multi-Class IP Networks using Utility Functions", PhD thesis, University of London.

[35] Reisslein, M., Rejagopal, S., Ross, K.W. (1998) "Guaranteeing statistical QoS regulated traffic" the multiple node case, IEEE conference on decision and control, December, 1998.

[36] Robbert, J.W., Mocci, J., Virtamo, J. (1996) "Broadband network teletraffic: performance evaluation and design of broadband networks", final report of action COST 242, Published by New York springer.

[37] S. Floyd and V. Jacobson, (1993) "Random early detection gateways for congestion avoidance", IEEE/ACM Transactions on Networking, pp.397-413.

[38] S. Floyd (2001), "Adaptive RED: An algorithm for increasing the robustness of RED's active queue management", AT&T Center for Internet Research.

[39] S. Floyd, (2000) "Recommendations on Using the Gentle Variant of RED", http://www.aciri.org/oyd 2000

[40] Saltzer, J. H., Reed, D. P., and Clark, D. D. (1984) "End-to-end arguments in system design: ACM Transactions on Computer Systems" Nov, 1984. .

[41] Schormans, J.A., Pitt, J.M., Scharf, E.M., Pearmain, A.J., and Philips, C.I. (2000), "Buffer Overflow Probability for Multiplexed on-off VoIP sources", Electronics Letters 2000,36, (6), pp.523-524 Private communication.

[42] Schormans, J.A., Pitts, J.M., Cuthbert, L.G., (1994), "Exact Fluid flow analysis of a single ON/OFF source feeding an ATM buffer, Electronics LettersZ", Vol 30, No. 14, July 1994, pp 1116-1117.