End-to-End QoS Evaluation of IP over LEO/GEO Satellites Constellations for FTP

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***Abstract*—This paper presents studies for the end-to-end QoS of IP over integrated terrestrial and Next Generation Satellite Network (NGSN) using FTP. We compare between LEO and GEO satellites constellations for the QoS parameters (i.e. delay, jitter, loss rate and throughput) of file transfer from a remote server in London and a remote client in Boston. We model the file transfer with multiple connections and file size variation according to Exponential and Pareto distributions respectively. We create the scenario with error model to simulate transmission loss environment using the NS-2 simulation software. A Differentiated Services (Diffserv) queue interface is placed in the server side to regulate the traffic flows across the narrow bandwidth of the satellite links. The results showed the performance evaluation and presented a good comparison of the QoS parameters involved in the data transfer across LEO and GEO satellites systems.**

Keywords-component; Quality of Service (QoS); IP over Satellite; Diffserv; FTP Application; Integrated Network

I. Introduction

The Next Generation Satellites Network (NGSN) plays a vital role in providing ubiquitous communications across the globe. Its unique characteristics like large coverage area, fast network deployment and native broadcasting/multicasting services extend the Internet connectivity to remote geographical area where terrestrial network is not available or not economical. With the latest standards from European Telecommunications Standards Institute (ETSI) [1] on digital video broadcasting like DVB-S/S2 [2, 3] for the forward channel and DVB-RCS [4] on the return channel, the satellite technology has been providing Internet broadband services at a competitive pricing rates, i.e. Tooway [5].

The future Internet will consists of integration of both terrestrial networks and satellite networks. Synchronize connection between the two networks is crucial in order to provide optimum end-to-end quality of service (QoS). The satellite networks are more prone to the transmission loss comparing to the terrestrial networks. In addition, the terrestrial networks have the upper hand in term of technology, bandwidth and speed (due to high speed and low bit error rate of optical fibre). The terrestrial network may leverage the data transfer over the satellite by adopting a control mechanism, i.e. Diffserv [6], to regulate and differentiate the traffic flows before being transmitted over the satellite. Unlike previous study on end-to-end QoS optimization of IP over satellites as in [7] which proposed an on-board processing (OBP) system, we introduce Diffserv queue interface in the terrestrial network to regulate and differentiate the multiple connections between server and clients. It provides scalability by simplifying the complexity functions such as traffic classification and traffic conditioning within the terrestrial edge routers [8, 9].

Previous related studies on end-to-end QoS of IP-Diffserv [10, 11, 12] only analyzed wired/wireless terrestrial networks without integrating with the satellites networks. None has done a top-down comparison on QoS parameters for data transfer using File Transfer Protocol (FTP) between LEO and GEO satellites constellations. The FTP is a common Internet protocol widely used to transfer large files (mainly referred as "elephants" [13]). It is built on TCP-based client-server architecture with separation of control and data connection between the client and server. In order to make file transfer through the Internet, a client has to establish a TCP connection to the server's well-known port 21. This connection is called the control connection which will remain open for the duration of the session. Then, the server responds with three digit status code in ASCII with an optional text message (connection negotiation dialog). If the connection establishment is successful, then a second connection is opened by the server from its port 20 to the client port (which is specified in the negotiation dialog) as required to transfer a file. Due to this two-port protocol structure, the FTP is considered as out-of-band as opposed to in-band protocol like HTTP [14].

This paper aims to evaluate and compare the QoS parameters (i.e. delay, jitter, loss rate and throughput) for Internet data transfer using FTP between integrated terrestrial-LEO and terrestrial-GEO networks. The NS-2 software package is used to simulate the internetworking scenarios for approximately one hour of simulation time. The rest of this paper is organized as follows. Section II describes the simulation configuration. Section III discusses the simulation results and analysis. Finally, section IV presents the conclusion and future works.

II. Simulation Configuration

The NS-2 simulation scenario is shown in Fig.1 which consists of a remote server, a remote client, a Diffserv queue

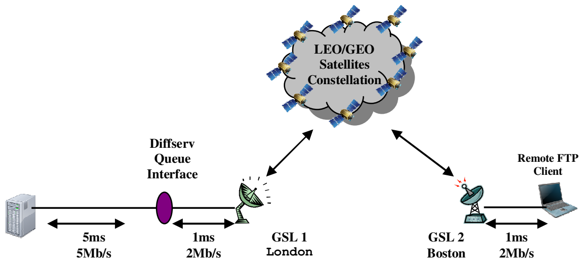


Figure 1. Simulation scenario.

interface, two ground stations to satellite link terminals (GSL) and the LEO/GEO satellites constellation. There are two different simulation scenarios used which are the terrestrial-LEO and terrestrial-GEO. Further details are described as follows.

A. *Satellites Network*

The NS-2 simulations configurations only differ in the satellites network parameters. The rest are the same for the whole simulations. We use Big LEO (i.e. 66 satellites) [15] and EuroSkyWay (i.e. 5 satellites) [16] as an example of LEO and GEO satellites constellation respectively. A remote server located in London (51.53^0 N, 0^0) transmits multiple TCP connections using FTP to a remote client located in Boston (42.3^0 N, 71.1^0 W). TABLE I shows the LEO and GEO parameters used throughout the simulations. Since the satellites network has high transmission errors [17], a random error model is introduced to simulate the characteristic. The error model produced three different bit-error-rates (BER) which are 10^{-7}, 10^{-6} and 10^{-5} for three different error scenarios.

TABLE I. LEO AND GEO SATELLITES PARAMETERS

Parameter	LEO Satellites	GEO Satellites
Altitude	780 Km	35786 Km
Planes	6	1
Satellites per plane	11	5
Inclination (degree)	86.4	0
Interplane separation (degree)	31.6	72
Seam separation (degree)	22	-
Elevation mask (degree)	8.2	8.2
Intraplane phasing	YES	YES
Interplane phasing	YES	NO
ISL per satellite	4	2
ISL bandwidth	25 Mb/s	25 Mb/s
Uplink/Downlink bandwidth	2 Mb/s	2 Mb/s
Cross-seam ISL	NO	NO
ISL latitude threshold (degree)	60	-

B. *Data Traffic Modeling for FTP*

The FTP connections vary randomly in term of average files sizes (i.e. 500 Kbytes, 1 Mbytes, 1.5 Mbytes and 2 Mbytes) and average new connection inter-arrival rate (i.e. between 1 connection/minute and 10 connection/minute) according to Pareto and Exponential distributions respectively. The TCP segment size is set to 576 bytes (i.e. 536 bytes of payload and 40 bytes of header) with maximum congestion window size of 30 packets. The main reasons for choosing small segment size and maximum congestion window are to accommodate many FTP connections within the 2 Mb/s of link bandwidth and also to reduce buffer overflow when the number or new connections increased. TABLE II shows the FTP connection parameters used in the simulations.

TABLE II. FTP CONNECTION PARAMETERS

Parameter	Value
FTP file size (bytes)	Model : Pareto Distrbution. Average: 500K, 1M, 1.5M, 2M bytes. Shape : 1.27
New connection inter-arrival rate (connection/minute)	Model : Exponential distribution. Average: 1, 2, 3, 4, 5, 6, 7, 8, 9, 10.
TCP type	New Reno
TCP packet size	576 bytes (536 bytes payload + 40 bytes header)

C. *Differentatised Services (Diffserv)*

Differentiated Services (Diffserv) is an Internet QoS architecture which is developed to resolve scalability problems and to provide preferential treatment to traffic flows based on class of service (CoS). The Diffserv queuing mechanism in the simulations used Random Early Detection (RED) queue and Time Sliding Window 3 Color Marker (TSW3CM) policer type which differentiate traffic flows based on 3 drop precedence (i.e. Green, Yellow and Red). Traffic flows classification will be based on the Committed Information Rate (CIR) and Peak Information Rate (PIR) which are set to 185 Kb/s and 190 Kb/s for a TCP connection. This setting is to allow 10 maximum average number of established TCP connections alive at a time with expected 90% - 95% link utilization (i.e. link bandwidth of 2 Mb/s).

Packets will be marked as Green if the flow rate within CIR, Yellow if the flow rate between CIR and PIR, and Red if the flow rate more than PIR. Red marked packets will be randomly dropped first followed by Yellow and Green packets respectively only if the buffer space exceeds minimum threshold. All packets will be dropped if the buffer space exceeds maximum threshold. All physical queue sizes used in both terrestrial and satellites networks are set to 100 packets. The minimum threshold size is set 30 packets which is equivalent to the TCP maximum congestion window while the maximum threshold is set to 90 packets. The reason is to allow buffer waiting space at a time equivalent to the TCP window size agreed upon connection establishment. Data packets will randomly dropped (i.e. drop probability equal to 0.1) if the buffer size between 30 and 90 packets and all data packets will be dropped (i.e. drop probability equal to 1) if buffer size more than that. Therefore 90% of the physical queue size is allocated for the data plane while 10% for the control plane.

III. RESULTS AND DISCUSSION

Each simulation is carried out for the duration of 1 hour of simulation time. The simulations are done 10 times (i.e. 10 average values of new connection inter-arrival time) for each FTP file size (i.e. 4 file sizes with average) in 3 different BER values. Therefore, the total numbers of repeated simulations are 240 times (i.e. for both terrestrial-LEO and terrestrial-GEO simulation scenarios). The simulation results and analysis will be divided into 4 QoS categories which are delay, jitter, loss ratio and throughput. In order to get better understanding of the following figures, we use the same reference symbol and annotation. There are in total of 12 colored lines on each graph which represent the QoS categories on 4 different FTP file sizes and 3 different BER values which are 10^{-7} (i.e. "□" symbol), 10^{-6} (i.e. "x" symbol) and 10^{-5} (i.e. "+" symbol).

A. *Average End-to-End Packet Delay*

The packet delay is measured by subtracting the packet received time at the client (t_r) to the packet sending time from server (t_s). The average delay (D) is measured by summing up all packets delays and then divided by the total number of successfully received packet (P_t) at the client side as shown in the following equation.

$$D(s) = \frac{\sum_{i=1}^{i=n} (t_r - t_s)_i}{P_t} \tag{1}$$

Fig. 2 shows that the average packet delay is proportional to the increment of average new connection per minute. The more new connection established per minute, the higher would be the delay. In addition, the delay also increased when the BER values increased from 10^{-7} to 10^{-5} due to the retransmission. Obviously, the delay values in Fig. 2 (b) are much higher than in (a) because of distinct difference in altitude between GEO and LEO satellites. Moreover, the propagation delay over GEO satellite more than 250 ms [18] as opposed to the LEO satellite which is more than 12 ms [6] depending on the hop count within the satellites network.

The delays steadily increased between 1 and 6 average new connection per minute. However, after 6 average new connections per minute, significant divergence could be seen between each flow of packet size with the maximum delay of 0.2724 and 0.3651 seconds (i.e. file size of 2 Mbytes) in LEO and GEO systems respectively. This is because of two main reasons which are the increment of queuing delay and the increment of packet retransmission. The queuing delay will increase when the number of incoming packet increase which will fill up the buffer space. The incoming packets of new flows keep on increasing regardless of the completion of previous flows. As the results, the packets incoming rates become more than the queue serving time. Besides that, the packet retransmission mainly happened because of early drop by Diffserv RED queue for the Red marked packets and also due to the packet drop in the satellite links.

B. *Average End-to-End Packet Jitter*

Packet jitter refers to the delays fluctuation or the delay difference between current received packet (D_c) and previous received packet (D_p). The jitter could be regarded as a vector variable because the positive value refers to the increment of current packet delay compared to the previous packet while the negative value refers to the decrement of current packet delay compared to the previous packet. Zero jitter means that the

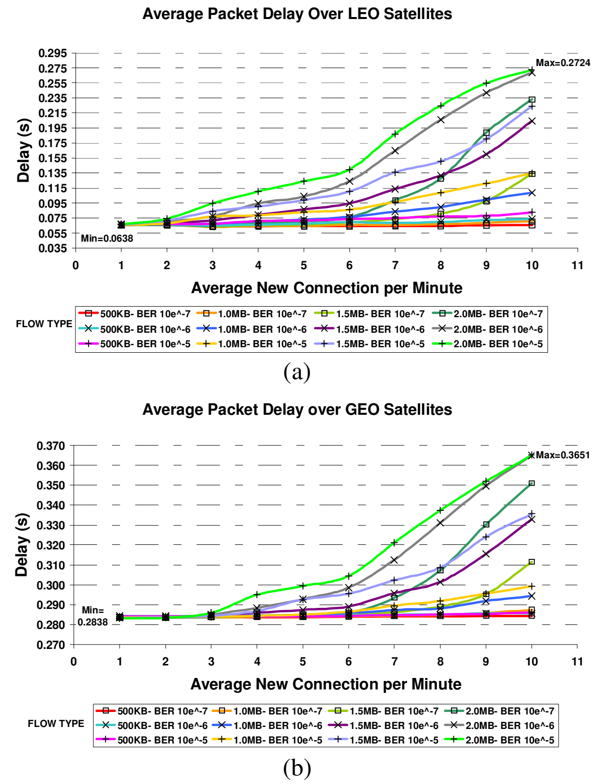


Figure 2. Average end-to-end packet delay.

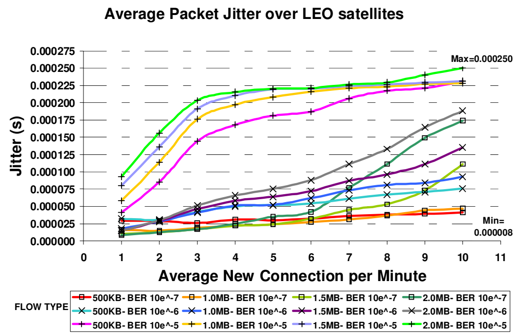
current packet delay is equal to the previous packet delay. The following equation shows the average jitter calculation.

$$J(s) = \frac{\sum_{i=1}^{i=n} (D_c - D_p)_i}{P_t - 1} \tag{2}$$

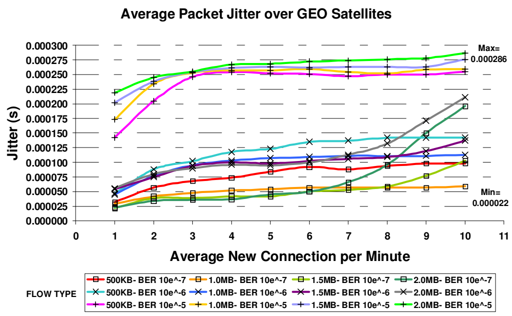
Fig. 3 shows that the average end-to-end packet jitter is proportional to the increment of average new connection per minute, average file sizes and BER. For BER values of 10^{-7} and 10^{-6}, steady increase of the average jitter could be seen between 1 and 6 of average new connection per minute and rapid increased for the subsequent connections. Higher file sizes has cause the TCP connections to remain active at longer time in order to complete the data transfer which eventually increase the influx of new connections at the queues. As the results, jitter variation could be seen when the queuing delay and packet loss retransmission increased. However, bigger gap in jitter could be seen for the flows with BER 10^{-5} which is the worst condition. This is due to the TCP time-out as the result of too many unsuccessful received packets at the client side.

C. *Average End-to-End Packet Loss Ratio*

Average packet loss ratio (L) refers to the ratio of total packet loss (P_l) over total transmitted packet from server to client (P_s). Equation (3) shows the loss ratio calculation.

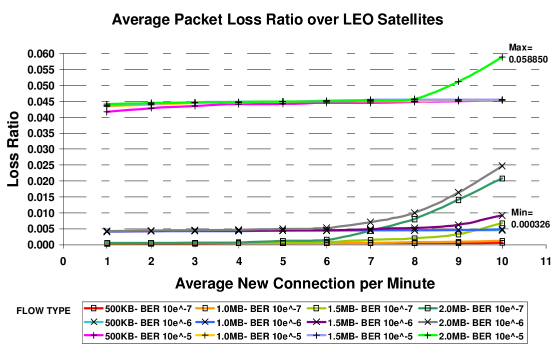


(a)

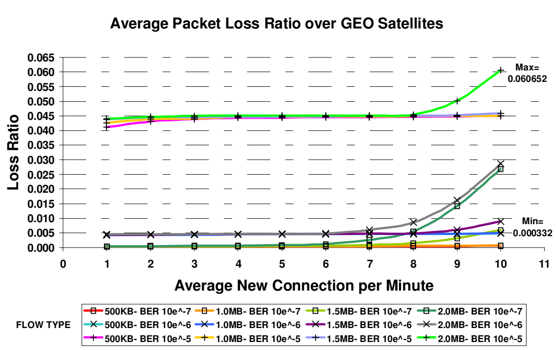


(b)

Figure 3. Average end-to-end packet jitter.



(a)



(b)

Figure 4. Average end-to-end packet loss ratio.

$$L = \frac{\sum_{i=1}^{i=n} (P_l)_i}{\sum_{i=1}^{i=n} (P_s)_i} \tag{3}$$

Fig. 4 shows that the packet loss ratio is proportional to the increment of average file sizes, average new connection per minute and BER. The loss rate values for all traffic flows over GEO satellites are slightly more than the one in LEO system. This mainly due to the higher round-trip-time (RTT) that cause the buffer space in most queues to fill up more quickly by the influx of new connections. In addition, the Diffserv regulate the flows by probabilistically drop packets when buffer size exceeds minimum threshold (i.e. influx rate > queue serving time). Besides that, the BER in satellite network also produce significant increment in loss rate especially above 10^{-6}.

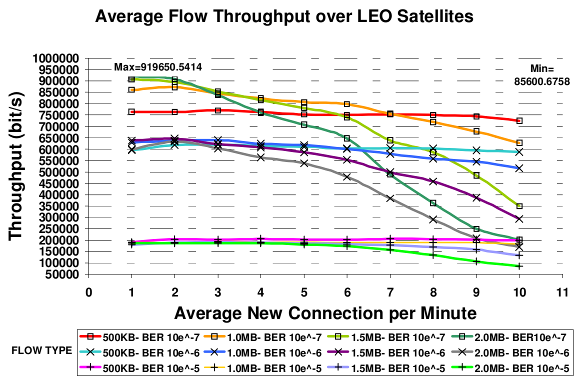
The minimum values could be seen at 1 average new connection, 500 Kbytes average file size and BER 10^{-7} which correspond to loss ratio of 0.000326 (i.e. LEO) and 0.000332 (i.e. GEO), while the maximum values are at 10 average new connection, 2 Mbytes average file size and BER 10^{-5} which correspond to loss ratio of 0.05885 (i.e. LEO) and 0.060652 (i.e. GEO). The loss rates are below 7 % under worst condition due to Diffserv QoS control and TCP reliable connection.

D. *Average End-to-End Flow Throughput*

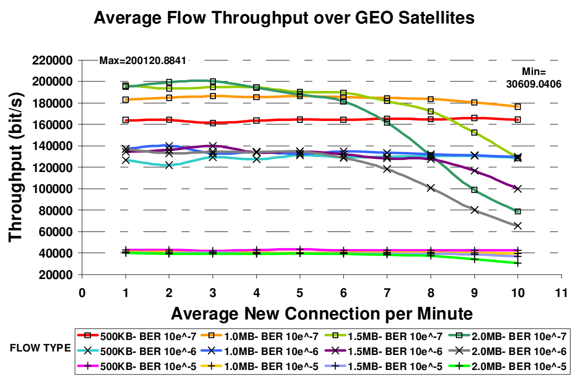
Flow throughput is calculated by dividing the total received packet bytes (P_b) over the duration of a FTP flow connection. The FTP flow duration calculated by subtracting the receiving time of last packet at the client (t_l) to the sending time of first packet of a flow at the server side (t_f). Then, the average flow throughput (T) is calculated by summing up all completed flow throughputs and divided by the total number of completed flows (f_t) as in (4).

$$T(bit/s) = \frac{\sum_{i=1}^{i=f_t} \left(\frac{P_b \times 8}{t_l - t_f} \right)_i}{f_t} \tag{4}$$

The throughput could be regarded as the conclusion of previous QoS parameters because they are closely related as shown in (4). Based on Fig. 5, the average flow throughput is inverse proportional to the increment of average new connection per minute and BER. The more competing flows exist in network, the lower would be the average throughput seen at the client side. However, the average throughputs are proportional to the average file sizes between 1 and 2 average new connection per minute. The throughputs steadily decline on the subsequent new connections and rapid decrement soon after 6 average new connections per minute. Apart from the BER values, this is because most of the FTP flows complete before the arrival of new connections (i.e. between 1 and 2) but takes long times to complete at subsequent average new connections especially after 6 average new connections due to the queuing delays and retransmission of packet loss.



(a)



(b)

Figure 5. Average end-to-end flow throughput.

IV. CONCLUSIONS AND FUTURE WORKS

This paper presented simulation studies to show top-down comparisons between terrestrial-LEO and terrestrial-GEO networks for the end-to-end QoS performance evaluations of FTP file transfers. The end-to-end QoS parameters (i.e. average delay, average jitter, average loss ratio and average flow throughput) are measured against the variation of average FTP file sizes (i.e. 500 Kbytes, 1.0 Mbytes, 1.5 Mbytes and 2.0 Mbytes), average new connection rate (i.e. between 1 and 10 connection/minute) and BER (i.e. 10^{-7}, 10^{-6} and 10^{-5}) for 1 hour of NS-2 simulation time. The average delay, jitter and loss ratio are proportional to the increment of average new connection per minute, average FTP file sizes and BER while the average flow throughput is vice-versa. Apart from the BER that significantly contribute to the increment of QoS parameters, the queuing delay, buffer size and scarce bandwidth limit the influx of new connections.

There still works remain for further studies. One of these is to achieve maximizing the bandwidth utilization on the satellite links by using load balancing method with multiple GSL on both server and client sides. This will involve multiple paths links from server to client. An admission control with Diffserv queue interface will be placed on the server side to regulate and control the flows paths over the satellites based on the current delay and throughput. This method will optimize the end-to-end QoS of multiservice applications like HTTP, FTP, video streaming and VoIP over the satellite links.

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