QoS Evaluation of Multiservice Applications Over Integrated Satellite-Terrestrial Networks

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Abstract—**This paper presents the QoS performance evaluation studies of IP over integrated terrestrial and Next Generation Satellite Network (NGSN) for HTTP web, file transfer, video streaming and VoIP applications. We compare the QoS parameters (e.g. delay, loss ratio and throughput) of the multiservice applications over Ka-Sat like satellite and the ITU-R standard Hypothetical Reference Digital Path (HRDP). We model the multiservice applications with multiple connections, different files sizes and connection durations variations. We simulate the network scenario with error model for the transmission loss environment using NS-2. A Differentiated Services (Diffserv) queue interface is used in the terrestrial network to regulate and differentiate the traffic flows while a priority queue is used as the satellite on-board-processing unit (OBP). The results showed a better top-down comparison of the QoS parameters involved in each application service across GEO satellite and the standard terrestrial digital data link.**

Keywords-component; QoS; IP over Satellite; Diffserv; OBP; Integrated Network; Multiservice Applications

I. INTRODUCTION

The launched of Ka-Sat to the space orbit (9^0 East above the equator) by Eutelsat Communications in December 2010 [1, 2] has marked a new milestone in the next generation satellite broadband industry. It has 70 Gbps of total throughput which will be channeled via 82 Ka-band spot beams on to different geographical areas stretching from the North Africa to southern Scandinavia and small part of Middle East. The Ka-Sat is expected to be operational in the second quarter of 2011 and has the notional capacity to serve up to two million households with triple-play services (e.g. Internet, Video on Demand (VoD) streaming and Voice over IP (VoIP)). One of the satellite broadband Internet Service Provider (ISP) that will use the Ka-Sat facilities is Tooway [3]. It is expected to deliver the high speed broadband services up to 10 Mbps download and 4 Mbps upload speeds.

Previous related studies on multiservice applications over the Digital Video Broadcast (DVB-RCS/S/S2) satellite broadband [4, 5, 6] systems only analyze the satellite network scenario without integration with the terrestrial network. In addition, the studies did not make comparison of the work done with any standard hypothetical reference in term of end-to-end quality of service (QoS) performance. We believe the future Internet broadband over satellite will comprise of both terrestrial and satellite networks and synchronize connection between both networks are vital in order to achieve optimum end-to-end QoS performance. Further comparison studies with the standard hypothetical reference is essential in order to know the potential of developed system so that further modification could be made to achieve better results. Unlike [7] which developed a complex OBP system for data traffic processing, we suggest an alternative approach by exploiting the terrestrial network capability to do the complexity functions such as traffic classification and traffic conditioning in order to relieve the satellite workload [8]. The reason is not only due to the higher satellite development cost but also because of the terrestrial networks have the advantage in term of technology, bandwidth and speed (e.g. high speed and low bit error-rate of optical fibre) compared to the satellite networks that have narrower bandwidth and prone to the transmission loss.

This paper aims to evaluate the QoS parameters of multiservice applications over the 10 Mbps of high speed satellite broadband using Ka-Sat like satellite system. We model the integrated terrestrial-satellite broadband services scenario in Network Simulator 2 (NS-2) software. A Diffserv queue interface is developed on the terrestrial network to regulate and differentiate the multiservice applications flows before crossing over the satellite network. In addition we proposed a simple priority queue with selective packets drop function as the satellite OBP. In order to make the NS-2 simulation more realistic, we create an error model which produced bit-error-rate (BER) in satellite links from the typical value of 10^{-7} [9] to the worst condition which is 10^{-6}. The applications traffics used in the simulations are HTTP/1.1 web, large file transfer using FTP, VoD streaming using MPEG-4 codec and bidirectional VoIP. Moreover, we also investigate the QoS parameters against multiple new connections rate, average response files sizes and BER variation using the standard ITU-R HRDP by replacing the satellite system with a single bidirectional optical fibre link. The results may give a better understanding of QoS parameters (e.g. delay, loss ratio and throughput) variations involved in multiservice applications across the satellite and HRDP systems.

The remainder of this paper is organized as follows: Section II explains the details of simulation configuration. Section III discusses the simulation results and analysis. Finally, section IV presents the conclusion and future works of this research.

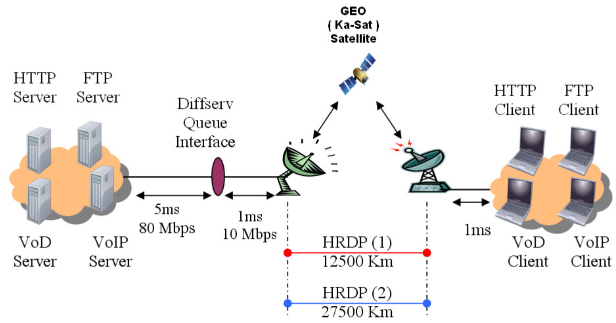

Figure 1. NS-2 simulation scenario.

II. SIMULATION CONFIGURATION

The NS-2 simulation network scenario is shown in Fig. 1 which consists of 4 remote servers, 4 remote clients, a Diffserv queue interface, two grounds to satellite links terminals (GSL), a Geosynchronous satellite (GEO) and a single bidirectional fibre optic link as the standard HRDP. There are actually three different network scenarios in Fig. 1 which are the terrestrial-satellite and terrestrial-HRDP. The main differences are only at the satellite and HRDP parameters while the rest network elements are the same. The next subsections explain the details of network elements parameters involved in the simulations.

A. *Satellite Network Configuration*

The satellite network used in the NS-2 simulations is based on the Ka-Sat like satellite system located at coordinate 9^0 east. There are 4 remote servers that transmit multiple TCP and UDP connections to 4 remote clients via 2 GSL located in London, UK (51.53^0 N, 0^0) and Athens, Greece (37.96^0 N, 23.72^0 E) respectively. We also introduced a random error model to simulate the satellite network transmission loss characteristics. The error model produced 2 different BER values which are 10^{-7} and 10^{-6} for 2 different error scenarios. TABLE I shows the satellite system parameters used throughout the simulations.

B. *ITU-R Hypothetical Reference Digital Path (HRDP)*

The International Telecommunication Union - Radio Communication Standardization Sector (ITU-R) has defined the HRDP in its S.521 document [10]. It is part of the Integrated Services Digital Network (ISDN) Hypothetical Reference Connection (HRX) which is defined in the ITU-T G.821 document [11]. Both HRDP and HRX defined the concept of satellite equivalent distance in terrestrial path. In addition, the HRDP and HRX specify the performance requirement of the main transmission segments for the end-to-end connection. HRX specified that the longest possible end-to-end connections between subscribers along the earth surface is 27500 Km. There are 3 basic segments identified by typical distances of portion in the end-to-end connection of HRX. The segments are referred as low, medium and high grade segments with allowable performance degradation of 30%, 30% and 40% respectively.

TABLE I. GEO SATELLITE PARAMETERS

Parameter	Value
Altitude	35786 Km
Coordinate	(0^0, 9^0)
ISL bandwidth	1 Gbps
Uplink / Downlink bandwidth	10 Mbps

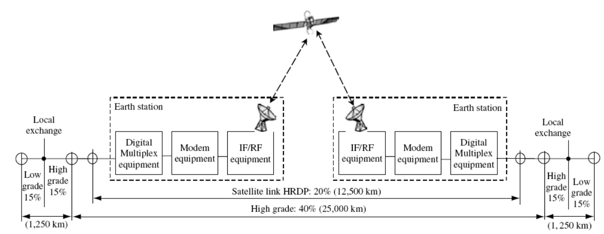

Figure 2. HRDP and HRX trasmission segments [12].

The HRDP is part of the high grade segment which represents the fixed satellite link with 12500 Km equivalent distance on earth. It consists of one terrestrial-satellite-terrestrial link with possibly more inter-satellite links in the space segment in the presence of many satellites. Fig. 2 shows the HRDP and HRX transmission segments as in [12].

We integrate the concept HRDP and HRX in the simulations by replacing the satellite segments with two single bidirectional terrestrial optical fibre links defined as HRDP(1) and HRDP(2) respectively as shown in Fig. 1. Therefore, there are 3 different network simulation scenarios in Fig. 1 which are using the satellite system, HRDP(1) and HRDP(2) respectively. The optical fibre specification is based on the standard optical-carrier-192 (OC-192) with transmission speed of 10 Gbps. Our objective is to study and compare the QoS variations involved in the satellite system and its standard equivalent distance of terrestrial link. The arguments that we used are "*what if*" the satellite system be replaced with the standard equivalent terrestrial ISDN links and "*how much*" the system effects the QoS performance.

C. *Multiservice Applications Traffic Modeling*

There are 4 applications services used in the NS-2 simulations which are the HTTP/1.1 web, large files transfer using FTP protocol, VoD streaming using MPEG-4 codec and bidirectional VoIP using GSM.AMR codec. Multiple connections are created for each type of application during the one hour of total simulation time. However, the average new connections inter-arrival rates are not uniform to all applications. The average new connection inter-arrival rate for HTTP web is varied between 1 and 5 per second while the FTP and VoIP are varied between 1 and 5 per minute. Only the VoD used 1 connection per minute for average new connection inter-arrival rate. Meanwhile, the new connection rate increment steps are uniform for HTTP, FTP and VoIP during the entire simulations. This means that when we increase the HTTP average new connection inter-arrival rate from 1/second to 5/second, we also increased the FTP and VoIP average new connection inter-arrival rate from 1/minute to 5/minute while the VoD average new connection inter-arrival rate remains at

1/minute. Detailed descriptions of the applications traffics used in NS-2 simulations are shown in the following subsections.

1) HTTP/1.1: The HTTP/1.1 web traffic used in NS-2 simulations is based on the Packmime-HTTP web application object that generates realistic synthetic web traffic [13]. However, we modified the average server reponse file size to be based on Pareto distribution with average value of 50 Kbytes. In addition, the average inter-arrival time for both request and response connections follow the marginal distribution which is a combination of modified fractional autoregressive integrated moving average (f-ARIMA) and Weibull distribution functions. The average new connection rates varies between 1 and 5 per second. Simplified discriptions of the complex equations of file size and new connection inter-arrival time distributions taken from NS-2 source codes are as follows.

The average server response file size is randomly generated using Pareto distribution based on average value (i.e. *avg_(x)*) of 50 Kbytes. Equation (1) shows the file size distribution function where *x* corresponds to the average file size. The *RNG* variable refers to the random number generator function that generates random numbers uniformly distributed between 0.0 and 1.0. The *S(x)* and *P* variables are the Pareto scale and shape parameters respectively. The *S(x)* variable formula as in (2) is based on average file size in (1) while the *P* is a constant value of 1.27.

$$f(x) = \frac{S(x)}{\lim_{n \to 1} RNG(n)^{(1/P)}} \quad (1)$$

$$S(x) = \frac{(P-1) \times avg\_(x)}{P} \quad (2)$$

The inter-arrival time distribution is based on (3) where *p* defined the f-ARIMA random distribution functions as in (4) and (5). The *shape* and *scale* variables as in (6) and (7) are the Weibull shape and scale parameters respectively. Both *shape* and *scale* are correlated with the average new connection rate (i.e. *R*) value that varies between 1 and 5. The *A* and *C* parameters are the sigma-epsilon and sigma-noise coefficients respectively while the *B* parameter is the f-ARIMA internal state coefficient. In addition, the *D*, *E*, and *F* are the Weibull coefficients while the *G* and *H* are the Gamma coefficients parameters. Detailed description of the following equations could be found in [13].

$$I_h(t) = -\log(1-p)^{(1/shape)} \times scale \quad (3)$$

$$p = f(y(t)) = \begin{cases} 0.5 & if \quad y(t) = 0 \\ \dfrac{1 + erf\left(\dfrac{y(t)}{\sqrt{2}}\right)}{2} & if \quad y(t) > 0 \\ \dfrac{erfc\left(\dfrac{-y(t)}{\sqrt{2}}\right)}{2} & if \quad y(t) < 0 \end{cases} \quad (4)$$

$$y(t) = A \times f\_ARIMA(Bt) + C \quad (5)$$

$$shape = \frac{2^{\left(D + \left(E \times \log\left(R/F\right)\right)\right)}}{1 + 2^{\left(D + \left(E \times \log\left(R/F\right)\right)\right)}} \quad (6)$$

$$scale = \frac{1}{R \times e^{\left(\log\left(G/1 + shape^{-1}\right) - H\right)}} \quad (7)$$

2) Large file transfer using FTP: The FTP application used in this study is for the Internet large file transfer with average file size value (e.g. *avg_(x)*) of 5 Mbytes based on Pareto distribution as shown in (1) and (2). Meanwhile, the average new connection rate varies between 1 and 5 connection/minute based on Exponential distribution shown in (8). The *avg_(t)* is the average inter-arrival time in 1 minute reference which are 60, 30, 20, 12 and 10 seconds corresponding to inter-arrival rate between 1 and 5 connection/minute respectively. The NS-2 scheduler used the the *avg_(t)* values to schedule the next packets transmission within the 1 hour of simulation time. The *RNG* is the random number generator function that generate numbers uniformly distributed between 0.0 and 1.0.

The FTP and HTTP web applications used TCP New Reno as the underlying transport protocol. The TCP segment size used is 1500 bytes (i.e. 1460 bytes payload + 40 bytes header) with maximum congestion window size of 30 packets.

$$I_e(t) = avg\_(t) \times (-\log(RNG(n))) \quad (8)$$
$$n \to 1$$

3) Video on Demand (VoD) streaming: The VoD streaming used MPEG-4 video encoder that generates 3 types of frames which are I frames, P frames and B frames. The I frames are the intra-coded frames that contains information of encoded still image and have the lowest compression rate compare to other type of frames. The P frames are the predictively coded frames that need information from previous I frames and/or P frames for encoding and decoding processes. The P frames could achieve higher compression rate than the I frames. Meanwhile, the B frames are the bidirectionally predictively coded frames that need information from the previous and following I and/or P frames for encoding and decoding processes. The B frames have the highest compression rate compare to the others. The I, P and B frames are generated using Time Expand Sample (TES) model in two phases. The first phase is to generate a time series of correlated random variables with uniform marginals [0, 1). The second phase is the inversion process of *background sequece* derived from the video sample trace files in NS-2. Further details of TES model could be read in [14].

The MPEG-4 video streaming in the NS-2 simulations used frame rate value of 24 frame/second and rate factor equal to 5 based on [14]. The average new connection created between server and client is only 1 connection/minute based on Exponential distribution shown in (8). Meanwhile the average streaming duration is 5 minute/connection based Pareto distribution as shown in (1) and (2). The new connection is created regardless of the completion of previous connection. The VoD used UDP as the underlying transport protocol with maximum transfer unit (MTU) of 1500 bytes.

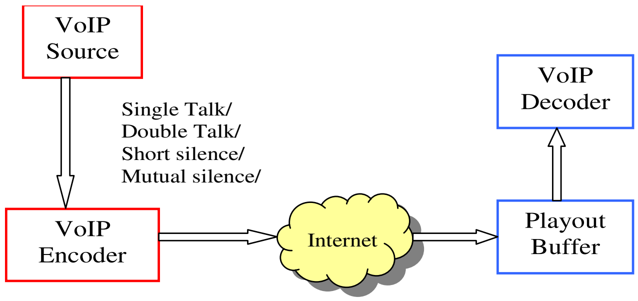

Figure 3. VoIP structure model in NS-2.

4) Voice over IP (VoIP): Bidirectional VoIP application is used bewtween the two communicating network elements in NS-2. At the source side, the VoIP encoder used GSM.AMR codec to encode the Voice Activity Detection (VAD) (e.g. talkspurt and silence activities) into 35 bytes of small packets before sending the data over the Internet channel using UDP. Upon receiving data from source, the VoIP decoder used the optimal non-causal playout buffer to pace their playout. The talkspurt and silence activities between the two conversation entities are model using modified Brady's model. The model contain 8 conversation states in which each state represent one of the following situations [15,16].

- Single talk: either one speaker is talking.
- Double talk: both speakers are talking at the same time.
- Short silence: either one speaker is silent.
- Mutual silence: both speakers are silent.

The average new VoIP connection rate between two conversation entities varies between 1 and 5 connection/minute based on the Exponential distribution shown in (8). Moreover, the average conversation duration is 10 minute/connection based Pareto distribution shown in (1) and (2). Similarly with other applications, the new connection is created regardless of the completion of previous connection. Fig. 3 shows the VoIP structure model used in the NS-2.

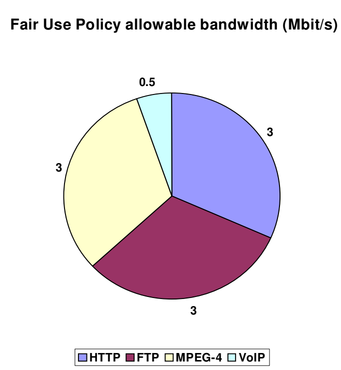

Figure 4. Bandwidth fraction for each traffic type.

D. Differentiated Services (Diffserv)

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 system in the NS-2 simulation used Random Early Detection (RED) queue type and the Time Sliding Window 3 Color Marker (TSW3CM) of policer type. The TSW3CM policer classifies traffic flows based on 3 drop precedence which are referred as *Green*, *Yellow* and *Red*. Meanwhile the RED queue consists of 1 physical queue and 3 virtual queues which correspond to the 3 drop precedence respectively. Traffic flows classification will be based on the Committed Information Rate (CIR) and Peak Information Rate (PIR). Packets will be marked as *Green* if the flow rate below CIR, *Yellow* if the flow rate between CIR and PIR, and *Red* if the flow rate more than PIR. The *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. Table II shows the Diffserv queue configuration used in NS-2. The total buffer size of a physical queue is 2000 packets with average packets size of 1500 bytes. The 3 virtual queues are virtually some fractions of the physical queue size which corresponds to the minimum threshold (*minTh*) and maximum threshold (*maxTh*). Assuming that 90% of the total buffer size used for user traffics, therefore the *maxTh* could be set equally to all traffic type which is 450 packets. The *minTh* is set less than *maxTh.*

In order to set the CIR and PIR for each type of traffic, we need to divide the maximum allowable bandwidth according to the fair use policy. The fair use policy is commonly used by the ISP to restrict the ways in which the network will be used by clients. Since we are studying the future services of Tooway satellite broadband services with 10 Mbps download and 4 Mbps upload speeds, therefore we used the 10 Mbps of total bandwidth as reference. Assuming that the ISP allows up to 95% (e.g. 9.5 Mbps) of the link utilization, therefore we divide the bandwidth to each traffic type as shown in Fig. 4. The bandwidth fraction corresponds to the PIR value of each traffic type. This does not mean that any traffic could not go beyond the PIR value. Any traffic could go beyond the PIR value by dynamically using other fraction of traffic type bandwidth as long as the total bandwidth follows the fair use policy. The CIR values are chosen to be less than the PIR values.

TABLE II. DIFFSERV PARAMETERS

Parameter	**HTTP**	**FTP**	**VoD**	**VoIP**
CIR (Mbps)	2.85	2.85	2.85	0.40
PIR (Mbps)	3.00	3.00	3.00	0.50
minTh (packet)	380	380	380	380
maxTh (packet)	450	450	450	450
Packet Drop Probability 1 (*Green*)	0.01	0.01	0.01	0.01
Packet Drop Probability 2 (*Yellow*)	0.05	0.05	0.05	0.05
Packet Drop Probability 3 (*Red*)	0.25	0.25	0.25	0.25

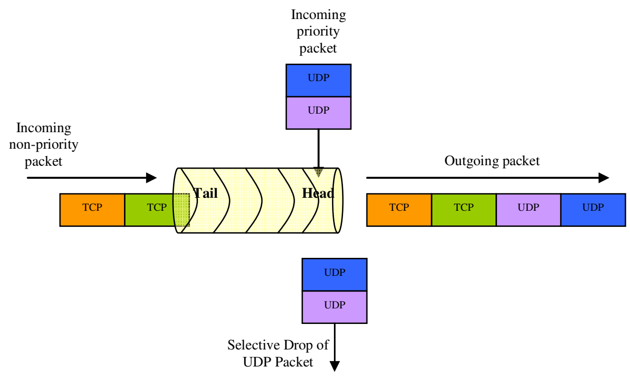

Figure 5. Priority queue model for the satellite OBP.

E. *Satellite On-Board Processing (OBP)*

The OBP refers to the satellite queuing management. It shows the capability of a satellite to manage the traffic flows variation and maintain the QoS at optimum level. We divide the traffic class into 2 categories which are *delay-sensitive* and *throughput-sensitive*. The HTTP web and FTP traffics used TCP with reliable connection. The TCP ensures all packet transmitted are successfully received and any packet loss will cause retransmission process. Higher packet loss will severely degrade the QoS. Therefore we categorized these traffics as *throughput-sensitive* which must be protected from being dropped by the queue. Meanwhile, the VoD (e.g. MPEG-4) streaming and VoIP traffics are the applications that sensitive to delay. Higher delay will severely degrade the QoS. Moreover, these traffics used UDP as the transport protocol which is non-reliable connection without retransmission function. Packet loss will no be retransmitted. Therefore, we categorized these applications traffics as *delay-sensitive*.

We model the OBP as a priority queue with selective packet drop function as shown in Fig. 5. The *throughput-sensitive* traffic will be queued from the tail of the queue while the *delay-sensitive* traffic will be queued from the head of the queue. This is done in order to make the delay-sensitive packets being served first by the queue. However, when the current queue size reach its maximum limit size (i.e. 2000 packets), then the *delay-sensitive* packets will be selectively dropped. Since the *delay-sensitive* packets could be divided into MPEG-4 video and VoIP packets, therefore either one of them will be dropped at a time. The queue will scan all packets in the buffer and counts the number of video and VoIP packets. If the video packets more than the VoIP packets in the buffer at that particular time, then the last video packet arrived in the queue will be dropped and vice versa.

III. SIMULATION RESULTS AND ANALYSIS

Each NS-2 simulation is carried out for approximately 1 hour of simulation time. The simulations are done 5 times for each connection rate values (e.g. *R* values between 1 and 5) in 2 different BER values (e.g. 10^{-7} and 10^{-6}). Therefore, the total numbers of repeated simulations are 30 times for terrestrial-GEO, terrestrial-HRDP(1) and terrestrial-HRDP(2) network scenarios. The simulation results and analysis are divided into 3 QoS categories which are delay, loss ratio and throughput. The QoS parameters are calculated based on each simulation output trace file using AWK programming script and then presented in the form of tables.

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

The packet delay is calculated by subtracting each packet received time at the client (t_r) to the packet sending time from the server (t_s). The average packet delay in second (D) is then calculated by summing all packet delays and then divided by the total number of successful received packets (P_t) at the client side as shown in (9).

$$D(s) = \frac{\sum_{i=1}^{i=n} (t_r - t_s)_i}{P_t} \quad (9)$$

The average end-to-end packet delay in second as shown in TABLE III, IV and V are proportional to the increment of average new connection from each traffic type. The average delays are steadily increased between 1 and 3 average new connection rate and then significantly increased on the subsequent connection rate with maximum delay achieved by flows with BER value of 10^{-6} in all systems and traffic types. When many connections are established per second or minute, the higher would be the end-to-end delay. This is mainly due to the increment queuing delay in most end-to-end data links in order to serve the increment incoming data rate. In addition, the average packet delay also increased when the BER increased from 10^{-7} to 10^{-6}. Moreover, the delay values are much higher in GEO system compare to HRDP(1) and HRDP(2) due to the distinct difference in altitude distance.

TABLE III. AVERAGE END-TO-END DELAY OVER GEO SATELLITE (S)

Traffic Type & BER		**Average New Connection**				
		1	***2***	***3***	***4***	***5***
HTTP	10^{-7}	0.2720	0.2721	0.2727	0.2763	0.2778
	10^{-6}	0.2730	0.2733	0.2744	0.2786	0.2798
FTP	10^{-7}	0.2739	0.2744	0.2751	0.2805	0.2847
	10^{-6}	0.2743	0.2757	0.2765	0.2838	0.2889
VoIP	10^{-7}	0.2653	0.2658	0.2669	0.2694	0.2711
	10^{-6}	0.2655	0.2665	0.2674	0.2706	0.2719
VoD	10^{-7}	0.2719	0.2722	0.2726	0.2749	0.2768
	10^{-6}	0.2724	0.2728	0.2734	0.2767	0.2783

TABLE IV. AVERAGE END-TO-END DELAY OVER HRDP(1) (S)

Traffic Type & BER		**Average New Connection**				
		1	***2***	***3***	***4***	***5***
HTTP	10^{-7}	0.0592	0.0593	0.0613	0.0686	0.0689
	10^{-6}	0.0597	0.0612	0.0635	0.0835	0.0836
FTP	10^{-7}	0.0597	0.0602	0.0618	0.0716	0.0727
	10^{-6}	0.0598	0.0622	0.0644	0.0858	0.0866
VoIP	10^{-7}	0.0522	0.0525	0.0538	0.0584	0.0601
	10^{-6}	0.0523	0.0530	0.0547	0.0659	0.0689
VoD	10^{-7}	0.0582	0.0585	0.0597	0.0657	0.0658
	10^{-6}	0.0594	0.0611	0.0630	0.0797	0.0803

TABLE V. AVERAGE END-TO-END DELAY OVER HRDP(2) (S)

Traffic Type & BER		Average New Connection				
		1	*2*	*3*	*4*	*5*
HTTP	10^{-7}	0.1087	0.1089	0.1107	0.1145	0.1162
	10^{-6}	0.1090	0.1094	0.1111	0.1221	0.1233
FTP	10^{-7}	0.1096	0.1109	0.1120	0.1167	0.1208
	10^{-6}	0.1096	0.1102	0.1114	0.1271	0.1301
VoIP	10^{-7}	0.1022	0.1024	0.1038	0.1069	0.1074
	10^{-6}	0.1024	0.1028	0.1046	0.1130	0.1149
VoD	10^{-7}	0.1080	0.1082	0.1093	0.1118	0.1136
	10^{-6}	0.1082	0.1090	0.1100	0.1219	0.1230

Apart from the distinct differences in propagation delays in all systems, the delay variation is caused by two main factors which are the increments of queuing delay in most links and the increment of packet retransmission of TCP flows. In addition, the rapid increased of delay after 3 average new connection rate is also due to the additional factor which is the early drop process by Diffserv queue for the *Red* marked packets. The incoming packets of new connection flows in all traffic type keep on increasing regardless of the completion of previous flows. When the traffics burst rate become more than the queue serving time, packets will be dropped and longer delay is needed to retransmit the TCP packets from server to client. Accumulation of TCP packets retransmission process will increase the TCP connection duration and subsequently increase the number of active connections in the end-to-end data links. As the results, this causes the global increment of packet delay. Besides that, the proposed priority queue managed to keep the delays for *delay-sensitive* traffics (i.e. VoD streaming and VoIP) lower than the *throughput-sensitive* traffics (i.e. HTTP web and FTP).

B. Average End-to-End Packet Loss Ratio

The packet loss ratio (L) defined the ratio of total packet loss (P_l) over the total transmitted packet from server to client (P_s) as shown in (10).

TABLE VI. AVERAGE PACKET LOSS RATIO OVER GEO SATELLITE

Traffic Type & BER		Average New Connection				
		1	*2*	*3*	*4*	*5*
HTTP	10^{-7}	0.000962	0.000980	0.001016	0.001032	0.001174
	10^{-6}	0.009938	0.010025	0.010196	0.010231	0.010292
FTP	10^{-7}	0.000983	0.001012	0.001098	0.001149	0.001248
	10^{-6}	0.010751	0.010803	0.010909	0.010955	0.011039
VoIP	10^{-7}	0.000031	0.000038	0.000052	0.000076	0.000099
	10^{-6}	0.000447	0.000460	0.000481	0.000502	0.000547
VoD	10^{-7}	0.001097	0.001110	0.001176	0.001228	0.001467
	10^{-6}	0.011526	0.011789	0.011943	0.012025	0.012054

TABLE VII. AVERAGE PACKET LOSS RATIO OVER HRDP(1)

Traffic Type & BER		Average New Connection				
		1	*2*	*3*	*4*	*5*
HTTP	10^{-7}	0.000070	0.000109	0.000131	0.000135	0.000200
	10^{-6}	0.001234	0.001238	0.001251	0.001305	0.001512
FTP	10^{-7}	0.000097	0.000123	0.000133	0.000138	0.000313
	10^{-6}	0.001298	0.001339	0.001341	0.001415	0.001555
VoIP	10^{-7}	0.000024	0.000029	0.000037	0.000055	0.000076
	10^{-6}	0.000049	0.000054	0.000075	0.000092	0.000135
VoD	10^{-7}	0.000120	0.000134	0.000138	0.000145	0.001122
	10^{-6}	0.001395	0.001432	0.001457	0.001491	0.002127

TABLE VIII. AVERAGE PACKET LOSS RATIO OVER HRDP(2)

Traffic Type & BER		Average New Connection				
		1	*2*	*3*	*4*	*5*
HTTP	10^{-7}	0.000103	0.000128	0.000131	0.000148	0.000201
	10^{-6}	0.001266	0.001285	0.001294	0.001372	0.001557
FTP	10^{-7}	0.000132	0.000136	0.000138	0.00018	0.000493
	10^{-6}	0.001343	0.001369	0.001383	0.001473	0.001640
VoIP	10^{-7}	0.000028	0.000033	0.000045	0.000069	0.000087
	10^{-6}	0.000054	0.000065	0.000082	0.000103	0.000151
VoD	10^{-7}	0.000136	0.00014	0.000155	0.000265	0.001222
	10^{-6}	0.001444	0.001491	0.001493	0.001522	0.002689

$$L = \frac{\sum_{i=1}^{i=n} (P_l)_i}{\sum_{i=1}^{i=n} (P_s)_i} \tag{10}$$

The average end-to-end packet loss ratio as shown in TABLE VI, VII and VIII is proportional to the increment of average new connection rate and BER values. The average packet loss ratio values for all traffic types in terrestrial-GEO are slightly more than the one in terrestrial-HRDP(1) and terrestrial-HRDP(2) systems. Apart from the BER factor, the average packet loss is also mainly because of the higher round-trip-time (RTT) that cause the buffer space in most queues to fill up more quickly by the burst of new traffic connections. In addition, the Diffserv queue regulates the traffic burst by probabilistically dropped packets when buffer size exceeds the minimum threshold (e.g. early packet drop process).

The minimum average end-to-end packet loss ratio for all systems could be seen in each traffic type at 1 average new connection rate and BER of 10^{-7}. Meanwhile, the maximum average loss ratio values are at 5 average new connection rate and BER 10^{-6}. The average packet loss ratio for VoD streaming traffic is higher than HTTP web and FTP traffics mainly due to the early packet drop process by the Diffserv and also selective packet drop by the priority queue. Besides that, the VoIP has the lowest average packet loss ratio because the traffic carries very small packet size and does not exceeds the fair use policy bandwidth fraction in most of the time.

C. Average End-toEnd Packet Throughput

The average end-to-end packet throughput (T) is calculated by dividing the total received packet (P_t) at the client side over the total duration of each traffic type. The value is then multiplied by 8 and divided by 1000 to get the value in Kbps. The application traffic duration is calculated by subtracting the receiving time of last packet at the client side (t_l) to the sending time of first packet from the server side (t_f) as shown in (11). The traffic duration is slightly less than 1 hour of simulation time because each traffic type starts a few seconds after the network simulation scenario setup is completed.

$$T(Kbps) = \frac{\sum_{i=1}^{i=n} (P_t)_i}{t_l - t_f} \times \frac{8}{1000} \qquad (11)$$

The average end-to-end packet throughput could be regarded as the conclusion of previous QoS parameters because the parameters are closely related as shown in (11). The average throughput is proportional to the total received packet variation and inverse proportional to the packet delay variation. Based on TABLE IX, X and XI, the average end-to-end packet throughput in Kbps is proportional to the increment of average connection rate, except for the VoD streaming traffic. The higher the average connection rate, the higher would be the P_t value in (11). The VoD streaming traffic is exceptional in this case because it uses only 1 average new connection/minute. The total transmitted packets remain almost the same during the entire simulation while the delay value (i.e. divisor in (11)) keeps on increasing when the average new connection rate of other traffic type increased. However, the average throughput values in all traffic flows are lower when the BER equal to 10^{-6} compared to the flows with BER 10^{-7} due to many packets loss.

The maximum average end-to-end packet throughput could be seen in terrestrial-HRDP(1) system at BER 10^{-7} while the minimum average end-to-end packet throughput could be seen in terrestrial-GEO system at BER 10^{-6}. The maximum average packet throughput is achieved at 5 average new connection rate for all traffics except the VoD streaming which is at 1 average new connection rate. The main reason other than the BER factor (e.g. as shown in TABLE VI, VII and VIII) that cause the lower average throughput in terrestrial-GEO system is also due to the higher end-to-end RTT (e.g. as shown in TABLE III, IV and V).

TABLE IX. AVERAGE THROUGHPUT OVER GEO SATELLITE (KBPS)

Traffic Type & BER		**Average New Connection 1**	**2**	**3**
HTTP	10^{-7}	343.5180	695.6649	1087.0275
	10^{-6}	343.3279	695.6377	1086.8133
FTP	10^{-7}	294.6074	677.6351	1098.4110
	10^{-6}	283.7430	675.9140	1087.3415
VoIP	10^{-7}	23.2299	43.0578	62.8611
	10^{-6}	22.5959	42.9940	62.3761
VoD	10^{-7}	1892.1931	1890.7961	1887.8901
	10^{-6}	1871.3099	1869.3138	1868.5890

Traffic Type & BER		**Average New Connection 4**	**5**
HTTP	10^{-7}	1475.3433	1951.7460
	10^{-6}	1468.5647	1924.8815
FTP	10^{-7}	1426.6242	1826.3049
	10^{-6}	1426.5160	1762.0156
VoIP	10^{-7}	85.8065	111.3068
	10^{-6}	85.4385	110.6742
VoD	10^{-7}	1887.3420	1887.2406
	10^{-6}	1868.5555	1867.3175

TABLE X. AVERAGE THROUGHPUT OVER HRDP(1) (KBPS)

Traffic Type & BER		**Average New Connection 1**	**2**	**3**
HTTP	10^{-7}	344.5814	696.3418	1087.5103
	10^{-6}	343.5607	696.0549	1087.2995
FTP	10^{-7}	300.2728	681.8100	1102.6363
	10^{-6}	299.9472	681.2453	1102.2178
VoIP	10^{-7}	23.5238	44.2878	63.2270
	10^{-6}	22.8913	43.3615	62.8749
VoD	10^{-7}	1897.7493	1893.8844	1892.6920
	10^{-6}	1896.3394	1890.4320	1888.7911

Traffic Type & BER		**Average New Connection 4**	**5**
HTTP	10^{-7}	1481.9453	2055.0792
	10^{-6}	1481.9407	2052.3165
FTP	10^{-7}	1437.2996	1877.6182
	10^{-6}	1436.4719	1877.3339
VoIP	10^{-7}	85.9698	112.3490
	10^{-6}	85.7534	110.7022
VoD	10^{-7}	1891.6254	1888.9905
	10^{-6}	1887.8191	1885.6149

TABLE XI. AVERAGE THROUGHPUT OVER HRDP(2) (KBPS)

Traffic Type & BER		**Average New Connection 1**	**2**	**3**
HTTP	10^{-7}	343.5757	696.0567	1087.1688
	10^{-6}	343.4940	696.0443	1087.0990
FTP	10^{-7}	299.1200	679.6220	1101.1449
	10^{-6}	298.5806	679.2519	1099.3751
VoIP	10^{-7}	23.3152	43.8924	62.9147
	10^{-6}	22.6584	43.1670	62.3838
VoD	10^{-7}	1893.0832	1891.4511	1891.4159
	10^{-6}	1888.8911	1887.8590	1885.4204

Traffic Type & BER		**Average New Connection 4**	**5**
HTTP	10^{-7}	1481.9078	2052.3576
	10^{-6}	1481.7544	2052.2844
FTP	10^{-7}	1432.4263	1872.1286
	10^{-6}	1431.7865	1870.5350
VoIP	10^{-7}	85.9508	111.6345
	10^{-6}	85.5202	109.6100
VoD	10^{-7}	1891.0840	1887.3282
	10^{-6}	1884.8885	1882.1653

IV. CONCLUSION AND FUTURE WORKS

This paper has presented the simulation studies to show top-down comparison between the GEO satellite system and its equivalent hypothetical terrestrial data link system (i.e. ITU-R HRDP) for the end-to-end QoS performance evaluation of multiservice applications (i.e. HTTP web, FTP, VoD streaming and VoIP). The end-to-end QoS parameters (i.e. average packet delay, average packet loss ratio and average packet throughput) are measured against the average connection rate and BER variation for 1 hour of NS-2 simulation time. The studies show that the QoS parameters variations are proportional to the increment of average connection rates and BER values. In addition, other parameters that contribute to the QoS parameters variations are the queuing delay and the buffer size. Moreover, the studies found that the GEO satellite system has lower end-to-end QoS performance for multiservice applications compared to the standard ITU-R HRDP terrestrial system mainly because of the distinct differences in round-trip-time (RTT). In addition, the priority queue with selective packet dropped scheme provides suitable QoS for the *delay-sensitive* traffics (i.e. VoD streaming and VoIP).

The future works aims to enhance the end-to-end QoS of the multiservice applications by using cross-layer method which will involve the transport and network layers. The global QoS degradation due to TCP retransmission processes could be reduced with TCP Performance Enhancement Proxy (PEP) method. The PEP will improve the TCP performance by using split connections and dynamic window resizing based on the available bandwidth. This method will significantly reduce the TCP RTT especially in GEO satellite system. Besides that, the network layer enhancement may involve load balancing method with multipath routing in order to optimize the bandwidth utilization. Moreover, an admission control with Diffserv queue system could be placed in the terrestrial network to regulate and differentiate the traffic flows based on current delay and throughput in order to reduce the satellite workload for data processing.

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